MultiVOIP™ 100

Voice / Fax over IP Networks

Voice/Fax Over IP Networks
Model MVP120

Proprietary Mode

User Guide

MultiTech® Systems

User Guide

MultiVOIP Series 100 Model MVP120
Product Number: S0000222 Revision A

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Patents

This product is covered by one or more of the following U.S. Patent Numbers: 6,219,708; 6,151,333; 5,757,801; 5,682,386. Other patents pending.

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World Headquarters

Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, Minnesota 55112
(763) 785-3500 or (800) 328-9717
Fax 763-785-9874
Tech Support (800) 972-2439
Fax-Back (763) 717-5888
Internet Address: **http://www.multitech.com**

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MultiVOIP™ 100

Voice / Fax over IP Networks

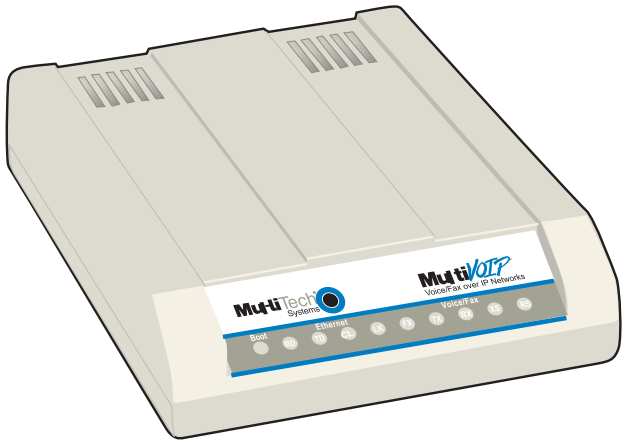
Chapter 1 - Introduction

MultiTech® Systems

Chapter 1 - Introduction to Multi-Tech's MultiVOIP

The MultiVOIP Series 100 (Model MVP120) allows analog voice and fax communication over a traditional data communications/data networking digital Internet. Multi-Tech's new voice/fax gateway technology allows voice and fax communication to ride, with no additional expense, over your existing communications Internet, which has traditionally been data-only. To access this free voice and fax communication, all you have to do is connect the MultiVOIP 100 to a phone or to your existing in-house phone system, and then to your existing Internet connection. Once configured, the MultiVOIP 100 then allows voice and fax to travel down the same path as your traditional data communications.

The MVP120 is designed with one FXO voice/fax channel, a 10 Mbps Ethernet LAN interface, and a command port for configuration. System management is provided through the command port using bundled Windows® software which provides easy-to-use configuration menus and a comprehensive Help system.



MultiVOIP 100 - Model MVP120

Preview of This Guide

This guide describes the MultiVOIP and tells you how to make the cable connections, install the software, and configure it. The information contained in each chapter is as follows:

Chapter 1 - Introduction to Multi-Tech's MultiVOIP
This chapter describes the MultiVOIP features, a typical application, and the front and back panels.

Chapter 2 - Installation
This chapter provides an overview of the installation process, describes the contents of the MultiVOIP package, diagrams the cabling instructions, and lists Telecom safety warnings.

Chapter 3 - Loading the Software
This chapter describes how to load the MultiVOIP software.

Chapter 4 - Configuring the Host MultiVOIP
This chapter describes how to configure your **Host.**

Chapter 5- Configuring the Client MultiVOIP
This chapter describes how to configure your **Client.**

Chapter 6 - Deploying the MultiVOIP Network
This chapter describes the duties of the VOIP administrator and how to connects the Client VOIPs.

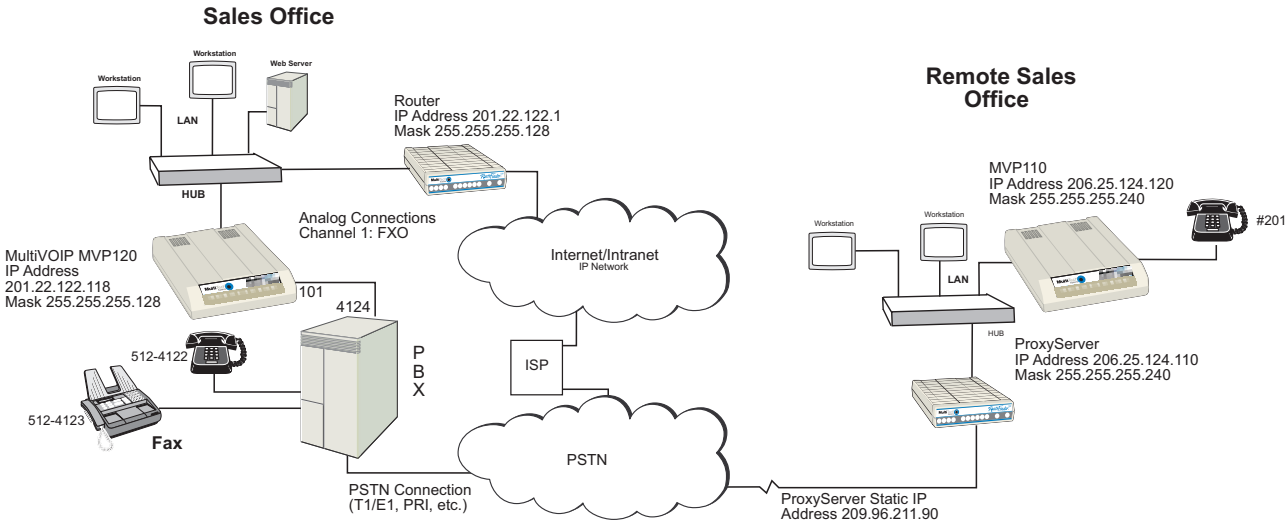
Chapter 7 - Using the MultiVOIP Software
This chapter describes how to use the software to change the configuration of the MultiVOIPs.

Chapter 8 - Remote Configuration and Management
This chapter describes how to change the configuration of a remote MultiVOIP. The chapter also describes typical client applications, such as Telnet and Web-based management.

Chapter 9 - Warranty, Service, and Technical Support
This chapter provides a statement of limited warranty, tells you how to get service for your MultiVOIP, and gives you information about the Multi Tech Systems, Inc. Web site.

Typical Application

Before VOIP (Voice Over IP) technology existed, a sales office used a data connection to the Internet **and** a voice connection to the public telephone network. Now, with VOIP, the two networks can be tied together. To accomplish this a Sales Office MultiVOIP is connected between the public telephone network and the data network as shown in the typical application below. The Sales Office MultiVOIP is set up as the master MultiVOIP and the MultiVOIP at the Remote Sales Office is set up as the client MultiVOIP. With this approach, the person at the Remote Sales Office can pick up the telephone and dial the Sales Office MultiVOIP and, after a second dial tone, can call anyone in the Sales Office or dial 9 for an outside line and then call anyone in the calling area as if they were in the Sales Office.



Typical VOIP Application

To set up the VOIP network, the MultiVOIP at the Sales Office is a MVP120 which has an FXO interface that allows it to be connected to an extension on the sales office telephone switch (PBX). The Ethernet port on the back of the MVP 120 provides the data connection to the data network.

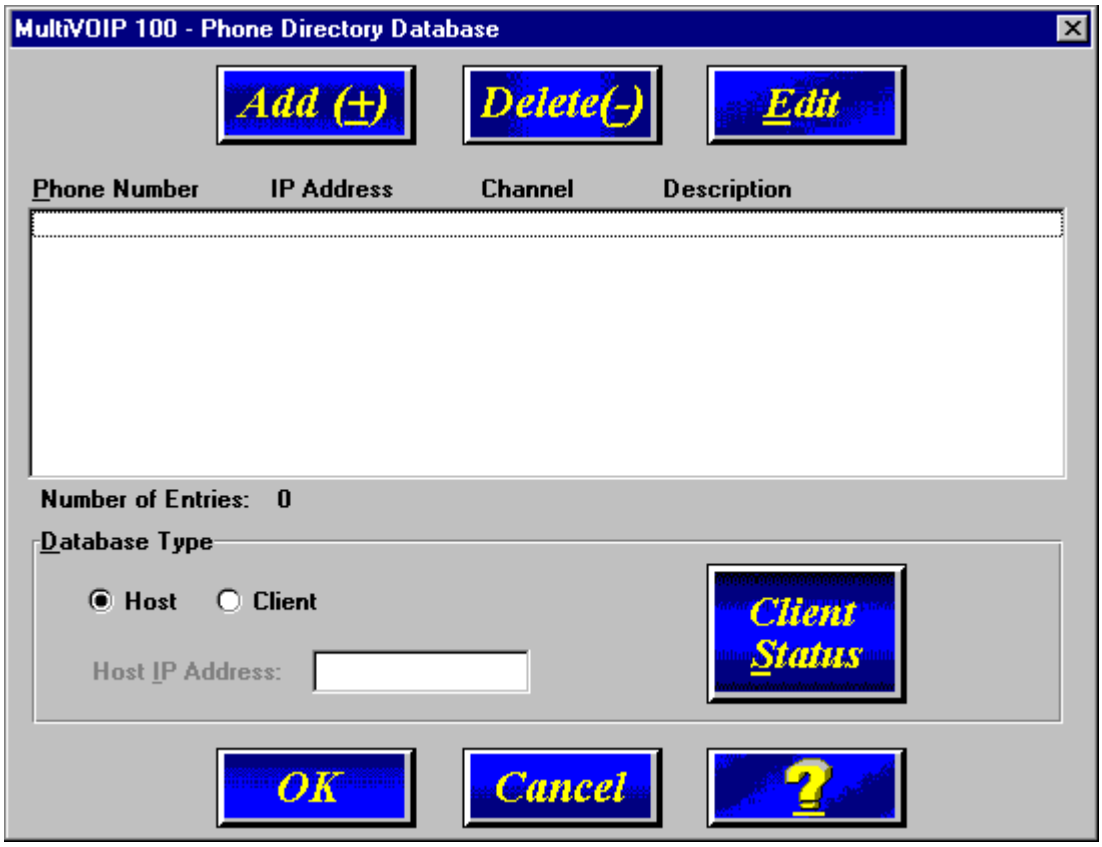
The MultiVOIP at the remote sales office is a MVP110 which has an FXS interface that allows a telephone to be connected. The Ethernet port on the back of the MVP110 provides the data connection to its data network.

To configure the Sales Office MultiVOIP, a PC is connected to the Command port on the back of the unit and configuration software is loaded onto the PC. The configuration software is based on a standard Windows Graphical User Interface (GUI), which simplifies your selection process to a single parameter group within a dialog box. For example, your LAN IP parameters are contained in a single dialog box. You can configure the IP address and mask for the MultiVOIP and the gateway address for the sales office router in the same dialog box. The Sales office MultiVOIP is then added to the data network.

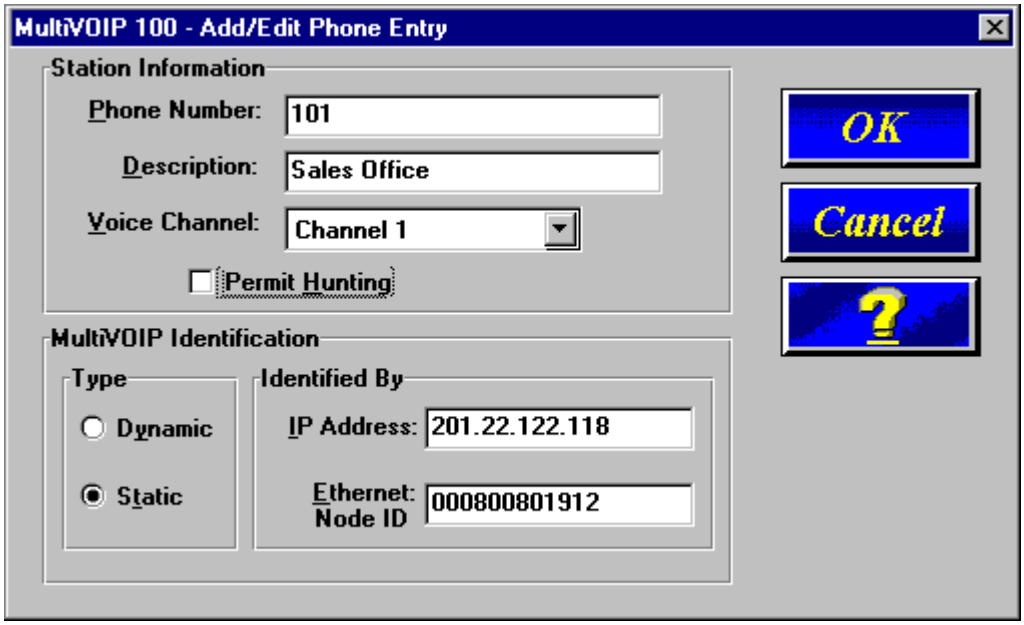
After your network configuration is complete, then you can develop your VOIP phone directory database. This database will be developed on the Sales Office MultiVOIP and sent out to the Remote Sales Office.

Phone Directory Database

The **Database Type** in the **Phone Directory Database** dialog box defines the Host to Client relationship. The Sales Office MultiVOIP in our typical application is set up as the Host, and the Remote Sales Office is set up as the Client. For the Remote Sales Office, you would need to know the IP address and the node ID or MAC address of the Sales Office MultiVOIP.



Now, to add a phone number to the phone directory database, you bring up the Add/Edit Phone Entry dialog box. Click **Add,** and the **Add/Edit Phone Entry** dialog box displays.

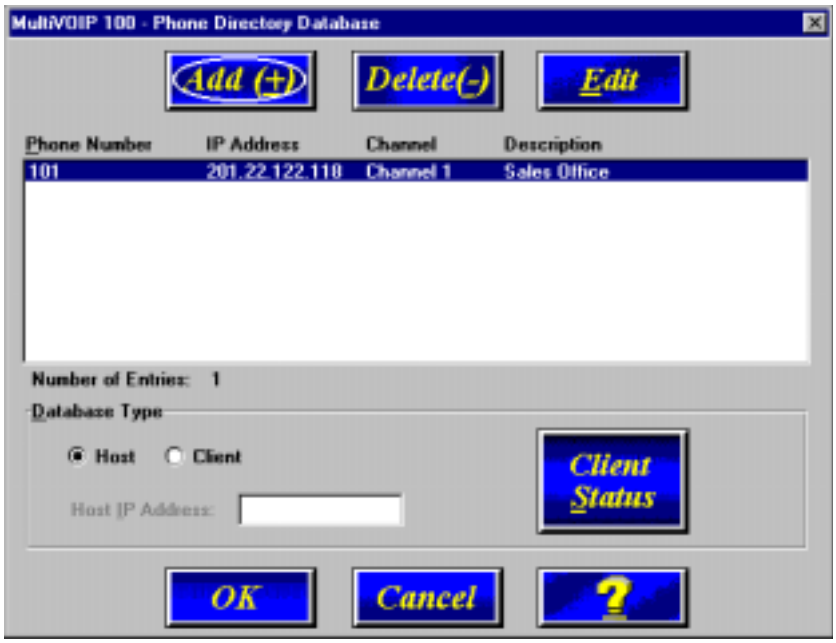


The **Station Information** group input the Phone Number, a Description (optional, but may be helpful if it is assigned to an individual or department; i.e., the Sales Office MultiVOIP), and Voice Channel of the unit, which in this case can only be Channel 1. The Permit Hunting option on the MultiVOIP 100 series does not apply.

The **MultiVOIP Identification** group defines the MultiVOIP for which you are assigning the phone number. The **Type** group (Dynamic or Static) defines the type of IP address that is used

for this MultiVOIP. The master MultiVOIP requires a static IP address so that the Client units know where the Host MultiVOIP is located in the network. The I**dentified By** group defines the IP address of the MultiVOIP for which the telephone number is being assigned. The Ethernet Node ID number is the 12-digit Node ID# (0008005xxxxx) that is assigned to every unit during the manufacturing process and cannot be changed. It is located on the ID plate on the back panel of the unit.

When you have entered this information in the **Add/Edit Phone Entry** dialog box, the new phone number is added to the Phone Directory Database.

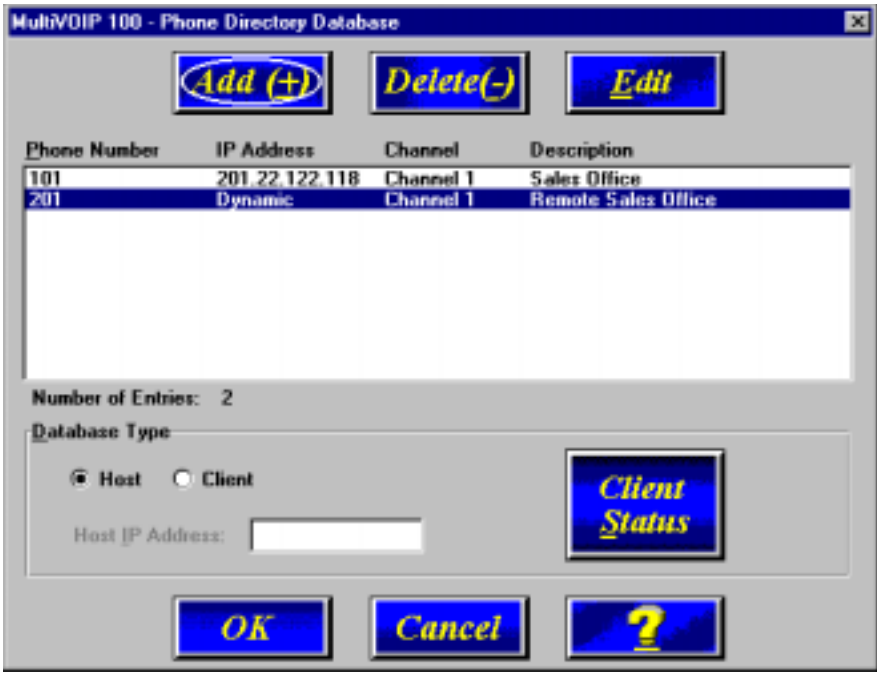


Now, to add the phone entry for the Remote Sales Office, you again bring up the **Add/Edit Phone Entry** dialog box and add the station information and identification of the Remote Sales Office MultiVOIP.

The **Station Information** is entered with the same reasoning as for the Host MultiVOIP. But, the **MultiVOIP Identification** for the Remote Sales Office Client unit depends on how it is used in its network. For instance, if the Client unit is behind a Proxy Server and the Proxy Server is providing the connection to the Internet and the Proxy Server is using dynamic

addressing (i.e., the ISP is assigning the IP addressing), then this Client MultiVOIP will use dynamic addressing, and the **IP Address** field in the **Identifed By** group will be greyed out (inactive). If the Proxy Server is using static addressing, then the **Type** group would be static and the **IP Address** field in the **Identified By** group would contain the IP address of the Proxy Server. If the Client MultiVOIP were to be connected directly to the Internet, then its addressing mode would have to be Static. If the Host unit is using static addressing, then the IP Address field has to contain the IP Address of the Client MultiVOIP.

When the Client unit is identified, then the second phone number is added to the phone directory database, and your two-unit VOIP network is setup.

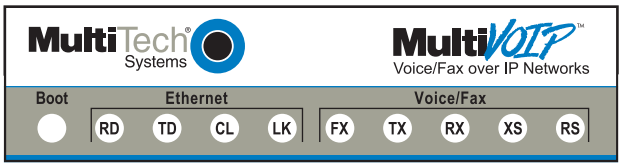


So, if a sales person in the Sales Office picked up the telephone and dialed extension 4124, the extension routes the telephone call to the VOIP network. After the second dial tone, the person would dial extension 201. The telephone at the Remote Sales office would ring and a telephone conversation is initiated just as if the Remote Sales office had been called over the regular telephone network.

Similarly, if a person at the Remote Sales office picked up extension 201 and dialed extension 101, the MultiVOIP at the Sales Office would generate a second dial tone and the person at the Remote Sales office would then dial any extension at the Sales Office, or the person could then dial a 9, for example, and get an outside line of the Sales Office Public Switched Telephone network. This person could then dial any telephone number, as if calling from the Sales Office.

Front Panel Description

The front panel has two groups of LEDs that provide the status of the Ethernet connection (Ethernet), the Voice/Fax channel, and an LED for the Boot status. The front panel is shown below, and a description of each LED follows.



MVP120 Front Panel

Boot

The Boot indicator lights when the MultiVOIP is booting or downloading setup.

Ethernet

RD Receive Data indicator blinks when packets are being received from the local area network.

TD Transmit Data indicator blinks when packets are being transmitted to the local area network.

LK Link indicator lights when the Ethernet link senses voltage from a concentrator or external device.

CL Collision indicator lights when a collision is detected on the Ethernet link.

Voice/Fax

FX Fax indicator lights when there is fax traffic on the voice/fax channel.

TX Transmit indicator blinks when voice packets are being transmitted to the local area network.

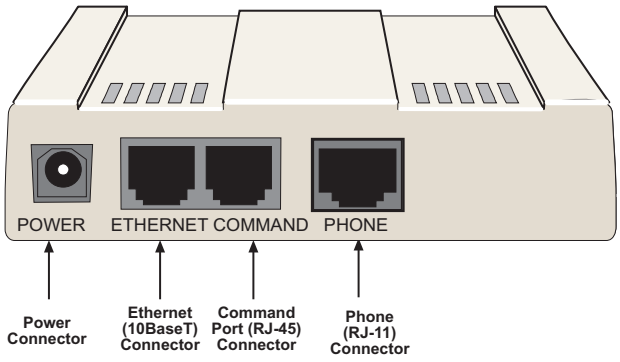
RX Receive indicator blinks when voice packets are being received from the local area network.

XS Transmit Signal indicator lights when the voice/fax channel is off-hook.

RS Receive Signal indicator lights when the voice/fax channel is ringing.

Back Panel Description

The cable connections for the MultiVOIP are made at the back panel. Connectors include Power, Command Port (RS232), Ethernet (10BASE-T), and the Voice/Fax Channel. The cable connectors are shown below and defined in the following groups.



MVP120 Back Panel

Power Connector

The Power connector is used to connect the power transformer to the MultiVOIP.

Ethernet Connector

The Ethernet connector is used to connect the MultiVOIP to a LAN using unshielded twisted cable. This connector is an RJ-45 jack.

Command Connector

The Command connector is used to configure the MultiVOIP using a PC with an available serial port and running Windows® software. The Command connector is an RJ-45 jack, and an adapter cable is provided to convert to a standard serial port DB9 female connector.

Phone Connector

Connects the Voice/Fax Channel to a PSTN or PBX/FXO port. This connector is an RJ-11 jack.

Specifications

- One 1 Meg by 32 byte at 70 nanosecond SIMM is 4 Mb DRAM
- Two Megabytes of flash memory

Ethernet Port

- Single Ethernet Interface--10BASE-T (twisted pair) keyed RJ-45 connector.

Command Port

- Single 19.2K bps asynchronous Command Port using an RJ-45 to DB9 cable with a DB9 female connector

Voice/Fax Channel

- One RJ-11 jack (FXO)

Power Supply

- Voltage - 115 VAC (Standard), 240 Volts AC (Optional)
- Frequency - 47 to 63 Hz
- Power Consumption - 4.3 Watts

Product

- Dimensions - 1.0" high x 4.3" wide x 5.6" deep
  (2.5 cm x 10.8 cm x 14.2 cm)
- Weight - 8 oz. (224 g)

MultiVOIP™ 100

Voice / Fax over IP Networks

Chapter 2

Installation

MultiTech® Systems

Chapter 2 - Installation

Overview of the Installation Process

The basic steps of installing your MultiVOIP network involve unpacking the units, connecting the cables, and configuring (Configuring is discussed in detail in Chapter 3) the units using the included management software (MultiVOIP Configuration). The recommended installation process includes four phases that, when completed, result in a fully functional Voice Over IP network.

Unpacking and Cabling the MultiVOIP

The MultiVOIP package includes the MultiVOIP 100, a printed Quick Start Guide, a system CD, a power supply, and a RJ-45 cable. If any one of these items is missing, please contact you sales representative. Detailed cabling directions are included in this guide.

Configuring the Host MultiVOIP

The VOIP administrator configures the MultiVOIP designated as the “Host” unit. This includes the assignment of a unique LAN IP address, subnet mask, and Gateway IP address. Once all connections have been made, the VOIP administrator configures the unit and builds the Phone Directory Database that will reside with the Host unit.

Configuring the Client MultiVOIPs

Unique LAN IP addresses, subnet masks, and Gateway IP addresses are assigned to the MultiVOIPs designated as “Client” units. When this is done, the Phone Directory Database option is set to Client, and the IP address of the Host MultiVOIP is entered. Once all client units are configured, the network is ready for deployment.

Deploying the VOIP Network

When the Client MultiVOIPs are sent to their remote sites, the remote site administrators need only to connect the units to their LAN and telephone equipment. A full Phone Directory Database (supplied by the Host MultiVOIP) will be loaded into their unit within minutes of being connected and powered up.

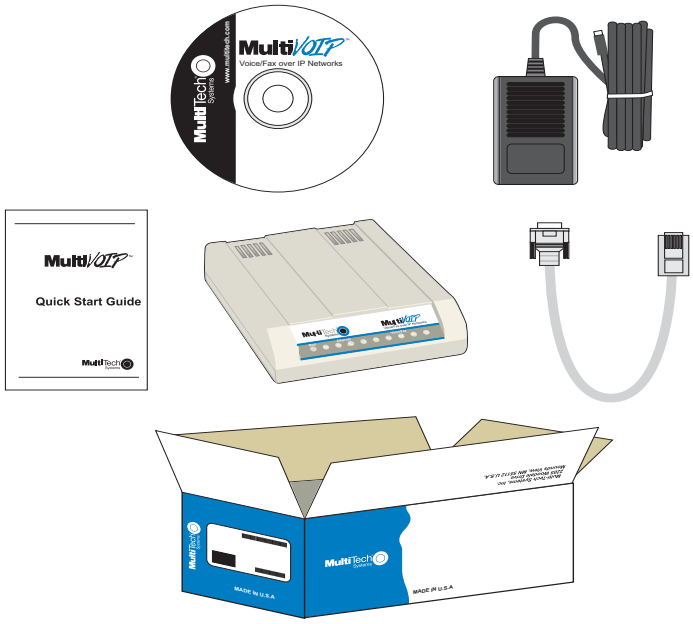
The final task of the VOIP Administrator is to develop the VOIP Dialing Directory based on the Phone Directory Database and telephone numbers of the interfacing telephone equipment. At this point, a VOIP user can call any person on the VOIP network.

Safety Warning Telecom

1. Never install phone wiring during a lightning storm.
2. Never install phone jacks in wet locations unless the jacks are specifically designed for wet locations.
3. This product is to be used with UL and cUL listed computers.
4. Never touch uninsulated phone wires or terminals unless the phone line has been disconnected at the network interface.
5. Use caution when installing or modifying phone lines.
6. Avoid using a phone during an electrical storm. There may be a remote risk of electrical shock from lightning.
7. Do not use the phone to report a gas leak in the vicinity of the leak.
8. To reduce the risk of fire, use only No. 26 AWG or larger telecommunications line cord.

Unpacking the MultiVOIP

Remove all items from the box. Check to make sure that all items are included.



Unpacking

Safety Warnings

1. A lithium battery on the circuit board provides backup power for the time keeping capability. The battery has an estimated life expectancy of ten years. When the battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for battery replacement.

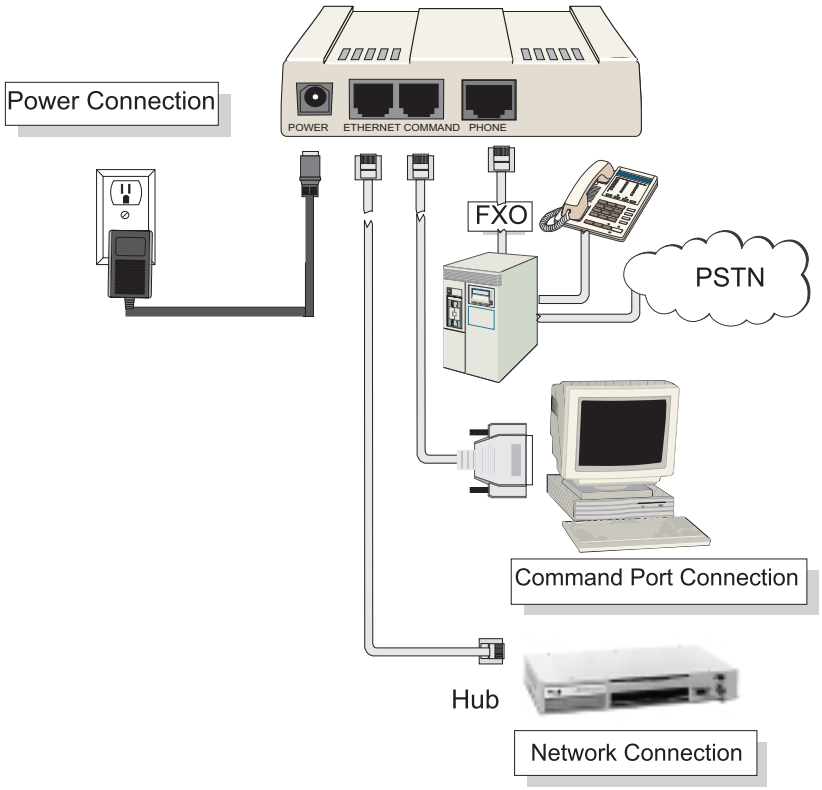
   Caution: Danger of explosion if battery is incorrectly replaced.

2. The Command and Ethernet ports are not designed to be connected to a Public Telecommunication Network. FXO can be connected to a PBX/FXO port or PSTN Network.

3. Use only the power source supplied with your product or an equivalent power source supplying the minimum power requirements.

Cabling the Host MultiVOIP

Cabling your MultiVOIP involves making the proper Power, Command Port, and Internet connections. The figure below shows the back panel connectors and the associated cable connections.



Cable Connections

Cabling Procedure

1. Using the supplied cable, connect the power supply to a live AC outlet, then plug the power supply into the MultiVOIP as shown above.

2. Connect the MultiVOIP to a PC using the RJ-45 to DB9 (female) cable provided with your unit. Plug the RJ-45 end of the cable into the **Command** port of the MultiVOIP and connect the other end to the PC serial port you are using.

3. Connect a network cable to the **Ethernet** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.

4. If you are connecting a PSTN or PBX/FXO port to your MultiVOIP, connect one end of an RJ-11 cable to the **Phone** connector on the back of the MultiVOIP and the other end to the PSTN or PBX/FXO port.

5. Turn on power to the MultiVOIP by moving the power switch on the right side panel to the ON position. Wait for the Boot LED on the MultiVOIP to go off before proceeding. This may take a few minutes.

   Note: If the Ethernet connection is not active, none of the LEDs will be on after booting. This unit does not have a Power LED.

MultiVOIP™ 100

Voice / Fax over IP Networks

Chapter 3

Loading the Software

MultiTech® Systems

Chapter 3 - Loading the Software

The following installation procedures do not provide every screen or option in the process of installing the MultiVOIP software. It is assumed that a technical person with a thorough knowledge of Windows and the software loading process is doing the installation. Additional information on the MultiVOIP software is provided in Chapter 4 and in the online Help.

The software loads a default configuration. Chapter Your Host MultiVOIP is configured with default parameters as involves software loading and configuration.

If you are installing a MultiVOIP behind a firewall, add the following UDP ports to your firewall.

900 Signaling Information

5000 Status Information

5004 RTP Packets

5005 RTCP Packets

Refer to your firewall user documentation to enter and open these ports.

1. Verify that your MultiVOIP 100 has been properly cabled, that the power is turned on, and that the **Boot** LED is off.

   Note: If the Ethernet connection is not active, none of the LEDs will be on after booting. This unit does not have a Power LED.

2. Insert the MultiVOIP 100 CD into your CD-ROM drive. The CD should start automatically. It may take 10 to 20 seconds for the Multi Tech CD installation window to display.

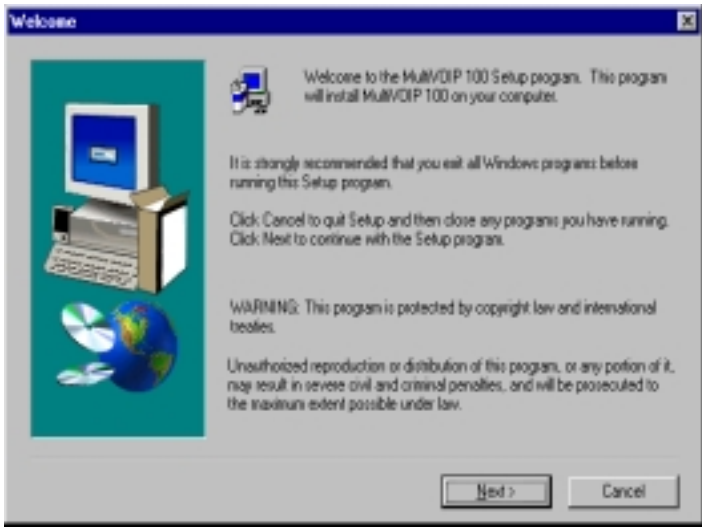
If the Multi-Tech Installation CD window does not display automatically, click **My Computer**, then right click the CD ROM drive icon, click **Open**, and then click the **Autorun** icon.

3. When the Multi-Tech Installation CD dialog box displays, click the **Install Software** icon.

   3a. Click **Install Software**.

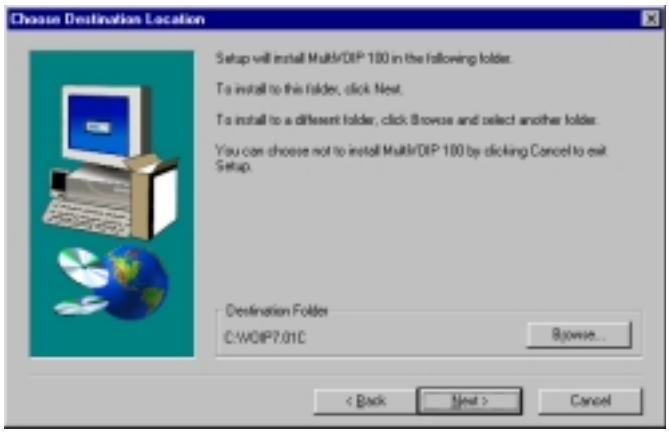
   3b. Click **Proprietary**, and then click **OK**.

4. The **MultiVOIP Setup** welcome window displays.


Welcome
Welcome to the MultiVOIP 100 Setup program. This program will install MultiVOIP 100 on your computer.
It is strongly recommended that you exit all Windows programs before running this Setup program.
Click Cancel to quit Setup and then close any programs you have running. Click Next to continue with the Setup program.
WARNING: This program is protected by copyright law and international treaties.
Unauthorized reproduction or distribution of this program, or any portion of it, may result in severe civil and criminal penalties, and will be prosecuted to the maximum extent possible under law.
Next >
Cancel

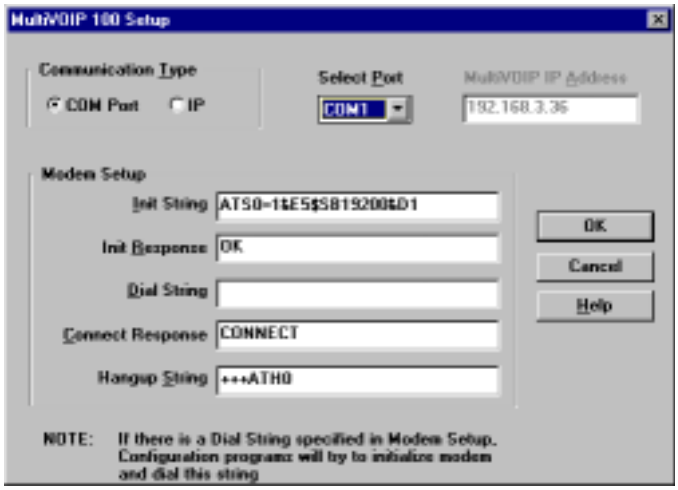
Press **Enter** or click **Next** to continue.

5. Follow the on-screen instructions to install your MultiVOIP software.

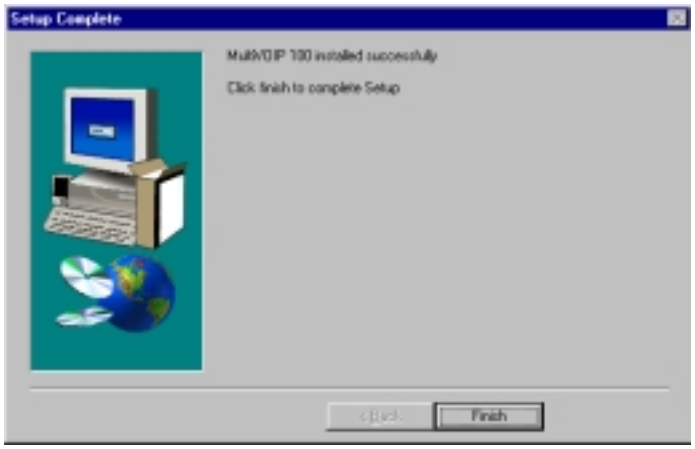

Choose Destination Location
Setup will install MultiVOIP 100 in the following folder.
To install to this folder, click Next.
To install to a different folder, click Browse and select another folder.
You can choose not to install MultiVOIP 100 by clicking Cancel to exit Setup.
Destination Folder
C:\VOIP7.01C
Browse...
< Back
Next >
Cancel

You can either choose the **Destination Location** of your MVP120 software or select the default destination by clicking **Next**. To choose your Destination Location, click the **Browse** button.

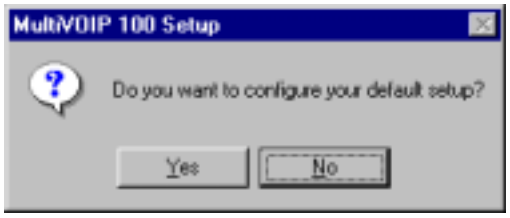
6. The **Setup** dialog box enables you to select the COM port of your PC that is connected to the Command port of the MVP120. From the **Select Port** list, select the COM port of your PC. Click **OK** to continue.



7. The **Setup Complete** dialog displays. Click **Finish**.



8. The following message displays. Click **No** to continue.



9. In the **Program** group, double-click **Upgrade Software**. The Proprietary software loads.

10. The following window displays. Click **Yes** to continue.

11. The **IP Protocol Default Setup** dialog box displays as you continue now with the configuring of your MultiVOIP, which is continued in Chapter 4.

MultiVOIP™ 100

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Chapter 4

Configuring the Host MultiVOIP

MultiTech® Systems

Chapter 4 - Configuring the Host MultiVOIP

The following steps provide instructions for configuring your MVP120. The configuration sequence includes IP Protocol default setup, Channel setup, and Phone Directory Database setup. The numbering of steps continues from the previous section.

12. The **IP Protocol Default Setup** dialog box:

The default **Frame Type** is **TYPE_II**. If this does not match your IP network, change the Frame Type by selecting SNAP from the **Frame Type** list. The available Frame Type choices are **TYPE_II** and **SNAP**.

13. In the **Ethernet** group, enter your unique LAN IP address in the **IP Address** box, and then enter the **Subnet Mask** and **Gateway Address** for your LAN in the corresponding boxes.

The IP address is your unique LAN IP address, and the Gateway address is the IP address of the device connecting your MultiVOIP to the Internet.

Click **OK** when you are finished.

14. The **Channel Setup** dialog box displays. It is used to define the voice/fax channel interface, voice coder, fax parameters, billing options, security settings, and regional phone parameters (tone pairs) for each channel. The **Interface** tab displays the interface, max dial digits, inter digit time, flash hook time, auto call, current loss, and ring count settings.

15. Check with your in-house telephone personnel to verify whether your local PBX dial signaling is **Pulse** or **DTMF** (tone). Select the regeneration option accordingly.

In the **Max Dial Digits** box, enter the maximum number of digits allowed when dialing a phone number. The default setting is 5. In the **Inter Digit Time** box, enter the maximum amount of time between dialed digits in milliseconds that the unit will wait before mapping the dialed digits to an entry in the Phone Directory Database. If too much time elapses between digits and the wrong numbers are mapped, you hear a rapid busy signal. If this happens, hang up and dial again. The default setting is **2** seconds.

In the **Flash Hook Timer** box, enter the amount of time the duration of flash hook signals output on the interface. The default setting is 600 milliseconds.

For FXO-to-FXO communications, you can enable a specific kind of **FXO Disconnect**: **Current Loss**, **Tone Detection**, or **Silence Detection**. Check with your in-house phone personnel to verify the type of disconnect to use. If **Current Loss** is checked, the VOIP will hang up when it detects a loss of current on the FXO port. For **Tone Detection**, select from the drop down list box one or two tones that will cause the line to disconnect. The person hanging up the call must then press the key or keys that produce those tones to hang up the call. For **Silence Detection**, select **One Way** or **Two Way**. Then set the timer for the number of seconds of silence before disconnect. The default value of 15 seconds may be shorter than desired for your application.

To dedicate the local voice/fax channel to a remote voice/fax channel, (so you will not have to dial the remote channel) select the **Auto Call Enable** check box. Enter the phone number of the remote VOIP in the **Phone Number** box.

16. The **Voice/Fax** tab displays the parameters for the voice coder, faxing, and DTMF gain, DTMF (Dual Tone Multi-Frequency) gain, voice coder, faxing, and advanced features such as Silence Compression, Echo Cancellation, and Forward Error Correction.

To change the voice coder, select the new voice coder entry from the **Voice Coder** list. If you changed the voice coder, ensure that the same voice coder is used on the voice/fax channel you are calling. Otherwise, you will get a busy signal.

Note: If **Automatic Coder** is selected, enter the bandwidth in the **Max Bandwidth** box. Check with your Network Administrator to determine available bandwidth.

You can set up the **Input Gain** and **Output Gain** so that the volume can be increased or decreased. Input gain modifies the level of the audio coming into the voice channel before it is sent over the Internet to the remote MultiVOIP. Output gain modifies the level of the audio being output to the device attached to the voice channel. Make your selections from the Input and Output drop-down lists in the **Voice** group. The valid range is +31dB to –31dB with a recommended/default value of **0**.

17. The **Fax** group enables you to select the maximum baud rate for faxes and adjust the fax volume. If you do not plan to send or receive faxes on a given voice/fax channel, you can disable faxes in the **Fax** group.

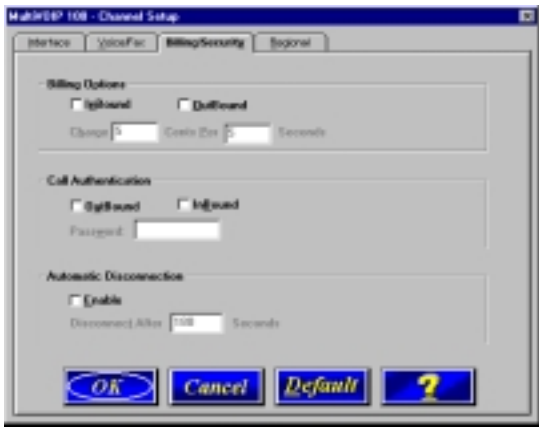
18. The **Miscellaneous** group lets you set voice/fax advanced features by clicking (checking) **Silence Compression**, **Echo Cancellation**, or **Forward Error Compression**.

The **Silence Compression** option defines whether silence compression is enabled for the selected voice channel. If enabled, the MVP120 will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth being used by the voice channel.

The **Echo Cancellation** options defines whether echo cancellation is enabled for the selected voice channel. If enabled, the MVP120 will remove echo, which improves sound quality.

The **Forward Error Correction (FEC)** option defines whether FEC is selected for the voice channel. This features allows some of the voice packets that were corrupted or lost to be recovered. FEC adds 50 % overhead to the total network bandwidth consumed by the voice channel.

19. The **Billing/Security** tab displays the parameters for billing options, call authentication, and automatic disconnection.



20. You can set up billing options for inbound and outbound calls by selecting them in the **Billing Options** group and then entering the charge in cents per number of seconds.

21. The **Call Authentication** option enables password protection for outbound and inbound calls on the selected voice/fax channel. If you enable password protection on inbound or outbound calls, you need to also enter a password of up to 14 numeric characters in the **Password** box.

22. The **Automatic Disconnect** option limits call duration to the number of seconds entered in the **Disconnect After** box. The default value of 180 seconds can be changed to any other value up to 65,535 (roughly 18.2 hours).

23. Click the **Regional** tab to change the Tone Pairs and to select your country/region from the **Country/Region** list.

The **Tone Pairs** group parameters change per your choice. Click **OK** when finished. The upgrade takes several minutes to complete.

24. The **Phone Directory Database** dialog box displays. You will build your personalized MultiVOIP Phone Directory in the following steps.

Note: The MultiVOIP configured as a "Host" will contain the host database. The host database has the phone numbers of all the MultiVOIP's available for communication on an IP network. This database is downloaded to each Client MultiVOIP as it comes online.

Click **Add** to begin building your phone directory database.

25. The **Add/Edit Phone Entry** dialog box displays.

In the **Station Information** group, enter the unique phone number of the local device connected to Channel 1 in the **Phone Number** box. For example phone number 101.

26. The **Description** is optional, but can be useful in associating the channel to the extension.

27. Enter the **Voice Channel** number corresponding to the phone number entered.

28. Click the **Permit Hunting** check box if you want calls to roll over to a second voice/ fax on an answering 2-, 4-, or 8-port client MultiVOIP when the first channel is busy.

    Note: Hunting will work between channels on the same MultiVOIP. However, hunting between channels on separate MultiVOIPs is not supported in proprietary mode.

    Note: The Host MultiVOIP must have a static IP address that the remote MultiVOIP can reference to obtain a downloaded copy of the host phone directory database.

29. In the **MultiVOIP Identification** group, enter the IP address of the Host MultiVOIP in the **IP Address** box. For example, 204.22.122.118. Then obtain the 12-digit Node ID# (0008005xxxxx) from the ID plate on the back panel of the MultiVOIP and enter this number in the **Ethernet Node ID** box. If the ID plate is missing or damaged, you can also Telnet to the MultiVOIP and, on the MultiVOIP Telnet Server menu enter **1** to advance to the Main Menu, then enter **3** for System Information where item 1 is the Ethernet Port Address you want to enter in the **Ethernet Node ID** box.

30. Click **OK** and you are returned to the **Phone Directory Database** dialog box, which now includes phone number 101 with its IP address, channel number, and description.

31. Click **Add** and the **Add/Edit Phone Entry** dialog box displays again.

32. Enter the phone number for the client MultiVOIP in the **Station Information** group **Phone Number** box. For example, 102.

33. Enter a description for the client MultiVOIP phone number for Channel 1 in the **Description** box.

Note: If the remote MultiVOIP is located behind a proxy server that uses a dynamically assigned IP address, select **Dynamic** (disabling Static IP Address) and leave the **IP Address** box blank. The Host MultiVOIP will learn the IP address when it is contacted by the client MultiVOIP.

34. Enter the IP address of the client MultiVOIP in the **IP Address** box in the **MultiVOIP Identification** group.

35. Click **OK** and you are returned to the **Phone Directory Database** dialog box, which now includes the second number and related information in the **Phone Number** list.

36. When you have finished, click **OK.** On the Main Menu, click **Download Setup.**

37. The **Save Setup** dialog box displays.

Click **OK** to proceed.

38. After the setup is written to the MultiVOIP, the unit is rebooted.

39. Verify that the **BOOT** LED on the MultiVOIP is off after the download is complete. This may take several minutes as the MultiVOIP reboots.

Note: If the Ethernet connection is not active, none of the LEDs will be on after booting. This unit does not have a Power LED.

40. You are returned to your MultiVOIP folder which is open and visible on your desktop.

At this time, your host MultiVOIP is configured. Proceed to the next section to configure the client MultiVOIPs.

MultiVOIP™ 100

Voice / Fax over IP Networks

Chapter 5
Configuring the Client MultiVOIPs

MultiTech® Systems

Chapter 5 - Configuring the Client MultiVOIPs

If you are installing a MultiVOIP behind a firewall, you need to add the following UDP ports to your firewall.

900 Q.931 Signaling Information	5004 RTP Packets
5000 Status Information	5005 RTCP Packets

Refer to your firewall user documentation to enter and open these ports.

1. Disconnect the PC from the command port of the Host MultiVOIP and connect it to the command port on the Client MultiVOIP.

2. From your desktop, click **Start | Programs I MultiVOIP I Upgrade Software**. The following dialog box displays.

3. Click **Yes**. The **IP Protocol Default Setup** dialog box displays.

The default **Frame Type** is **TYPE_II**. If this does not match your IP network, select the Frame Type from the **Frame Type** list. The Frame Type choices are **TYPE_II** and **SNAP**.

4. In the **Port Address** group, enter the **IP Address** and **IP Mask**. In the **Gateway Address** group, enter the gateway IP address for the client unit.

   The IP address is your unique LAN IP address, and the Gateway address is the IP address of the device connected to the Internet/Intranet.

   Click **OK** when you are finished. The **Channel Setup** dialog box displays.

5. The **Channel Setup** dialog box defines the voice coder, fax parameters, and regional telephone parameters for the voice channel.

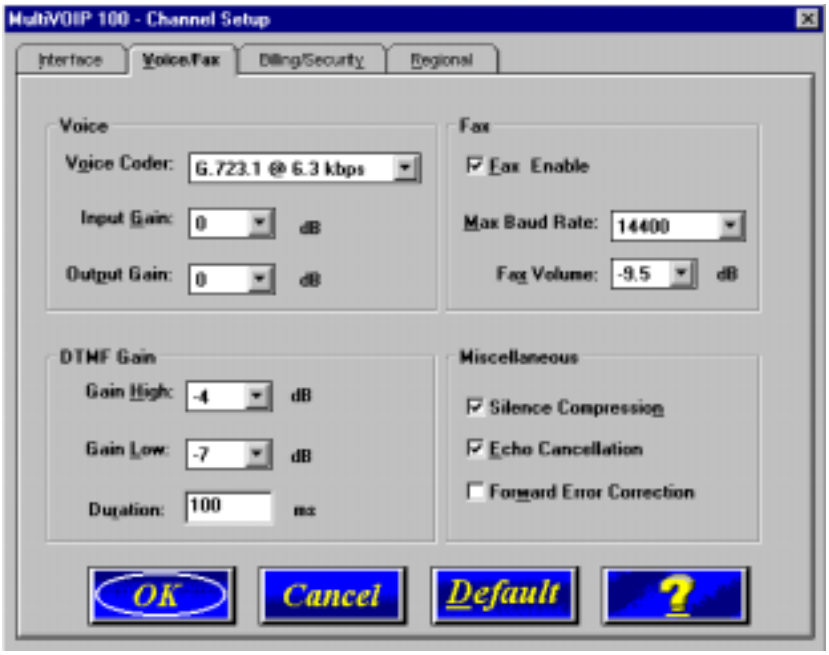
6. Check with your in-house telephone personnel to verify whether your local PBX dial signaling is **Pulse** or **DTMF** (tone). Select the regeneration option accordingly.

   In the **Max Dial Digits** box, enter the maximum number of digits allowed when dialing a phone number. The default setting is 5. In the **Inter Digit Time** box, enter the maximum amount of time between dialed digits in milliseconds that the unit will wait before mapping the dialed digits to an entry in the Phone Directory Database. If too much time elapses between digits and the wrong numbers are mapped, you hear a rapid busy signal. If this happens, hang up and dial again. In the **Flash Timer** box, enter the time, in milliseconds, for the duration of flash hook signals output on the interface.

   For FXO-to-FXO communications, you can enable a specific kind of **FXO Disconnect: Current Loss**, **Tone Detection**, or **Silence Detection**. Check with your in-house phone personnel to verify the type of disconnect to use. If **Current Loss** is checked, the VOIP will hang up when it detects a loss of current on the FXO/phone port. For **Tone Detection**, select from the drop down list box one or two tones that will cause the line to disconnect. The person hanging up the call must then press the key or keys that produce those tones. For **Silence Detection**, select **One Way** or **Two Way.** Then, in the **Silence Timer** box, enter the number of seconds of silence before disconnect. The default value of 15 seconds may be shorter than desired for your application.

   To dedicate the local voice/fax channel to a remote voice/fax channel, (so you will not have to dial the remote channel) check the **Auto Call Enable** check box. Enter the phone number of the remote VOIP in the **Phone Number** box.

7. The **Voice/Fax** tab displays the parameters for the voice coder, faxing, DTMF (Dual Tone Multi-Frequency) gain, and the miscellaneous advanced features **Silence Compression**, **Echo Compression**, and **Forward Error Correction**.



To change the voice coder, select the new voice coder entry from the **Voice Coder** list.

If you changed the voice coder, ensure that the same voice coder is used on the voice/fax channel you are calling. Otherwise, you will always get a busy signal.

Note: If **Automatic Coder** is selected, enter the bandwidth in the **Max Bandwidth** box. Check with your Network Administrator to determine available bandwidth.

8. The **Fax** group enables you to select the maximum baud rate for faxes and adjust the fax volume. If you do not plan to send or receive faxes on a given voice/fax channel, you can disable faxes in the **Fax** group.

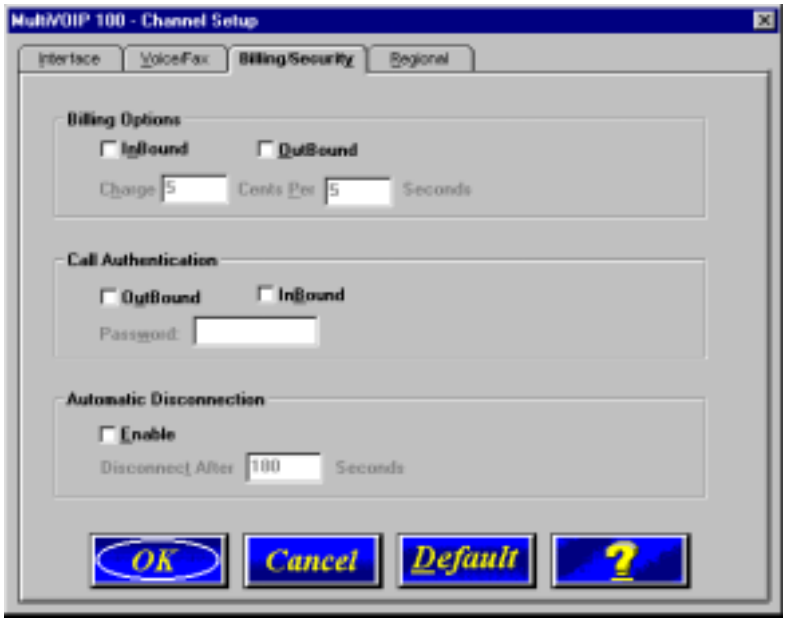
   The **Miscellaneous** group lets you set voice/fax advanced features by clicking (checking) **Silence Compression**, **Echo Cancellation**, or **Forward Error Compression**.

   The **Silence Compression** option defines whether silence compression is enabled for the selected voice channel. If enabled, the MVP120 will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth being used by the voice channel.

   The **Echo Cancellation** options defines whether echo cancellation is enabled for the selected voice channel. If enabled, the MVP120 will remove echo, which improves sound quality.

   The **Forward Error Correction (FEC)** option defines whether FEC is selected for the voice channel. This features allows some of the voice packets that were corrupted or lost to be recovered. FEC adds 50 % overhead to the total network bandwidth consumed by the voice channel.

9. The **Billing/Security** tab displays the parameters for billing options, call authentication, and automatic disconnection.



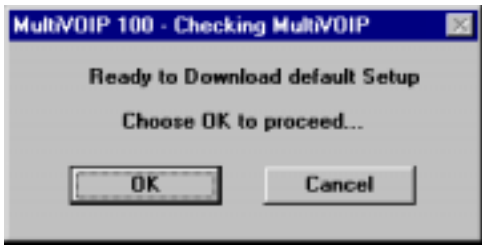
10. You can set up billing options for inbound and outbound calls by selecting them in the **Billing Options** group and then entering the charge in cents per number of seconds.

11. The **Call Authentication** option enables password protection for outbound and inbound calls on the voice/fax channel. If you enable password protection on inbound or outbound calls, you need to also enter a password of up to 14 numeric characters in the **Password** box.

12. The **Automatic Disconnect** option limits call duration to the number of seconds entered in the **Disconnect After** box. The default value of 180 seconds can be changed to any other value up to 65,535 (roughly 18.2 hours).

13. Click the **Regional** tab to select your specific country/region from the **Country/ Region** list or to change the **Tone Pairs**.

The **Tone Pairs** group parameters change per your selections. Click **OK**. The following dialog box displays.



14. Click **OK**.

After the upgrade is complete, click **Start | Programs| MultiVOIP | MultiVOIP Configuration.** On the main menu, click **Phone Book** to display the **Phone Directory Database** dialog box.

In the **Database Type** group, click the **Client** option. The **Host IP Address** box becomes active.

15. Enter the IP address of the host MultiVOIP in the **Host IP Address** box.

16. Click **OK** and you are returned to the main menu.

17. Click **Download Setup** to write the new configuration to the client unit. The **Save Setup** dialog box displays.

18. Select the **Save Current Setup as User Default Configuration** check box and click **OK**. After the setup is written to the MultiVOIP, the unit reboots.

19. Verify that the **Boot** LED on the MultiVOIP is off after the download is complete. This may take several minutes as the MultiVOIP reboots.

20. You are returned to the main menu.

    Your MultiVOIP is operational at this time.

Repeat this procedure for each of the client units. When all clients have been configured, proceed with deploying the VOIP network.

MultiVOIP™ 100

Voice / Fax over IP Networks

Chapter 6
Deploying the VOIP Network

MultiTech® Systems

Chapter 6 - Deploying the VOIP Network

Deploying the VOIP network involves the VOIP Administrator developing the VOIP Dialing Directory and deploying the pre-configured client MultiVOIPs to their remote sites. The remote site administrators need only connect power to the pre-configured MultiVOIP, connect it to their Ethernet LAN and predefined telephone equipment, and then wait for the phone directory database to be downloaded.

Perform the following procedure to deploy your VOIP network.

VOIP Administrator

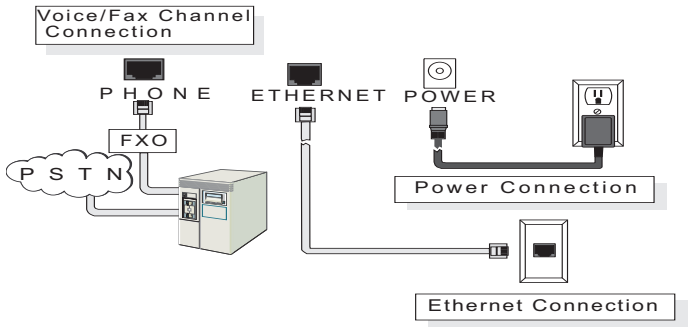
1. Create your VOIP Dialing Directory based on your Phone Directory Database for the numbers to connect the MultiVOIPs to your VOIP network and the telephone extension number you need to connect to the Voice/Fax channels. A sample VOIP Dialing Directory is provided below for your consideration and use.

VOIP Dialing Directory		
To call from	**Call Process**	**Dialing Sequence**
Corporate Office to Remote Branch Office	Pick up any phone and dial an extension number (e.g., 4124). Upon hearing the second dial tone, dial the Remote Branch Office MultiVOIP (301).	4124 301
Remote Branch Office to Corporate Office	Pick up telephone and dial 101. Upon hearing the second dial tone, dial 4124.	101 4124

2. Send the client MultiVOIPs to their remote sites.

Connecting the Client MVP120

1. Connect one end of the power supply to a live AC outlet and connect the other end to the **Power** connection on your MultiVOIP.



Remote Site Cable Connection

2. Connect a network cable to the **ETHERNET** (RJ-45) connector on the back of your MultiVOIP.
3. If you are connecting a PSTN or PBX/FXO port, connect one end of an RJ-11 cable to the **Phone** connector on the back of the MultiVOIP and the other end to the PSTN or PBX/FXO port.
4. Turn on power to the MultiVOIP by moving the ON/OFF switch on the right side panel to the ON position. Wait for the **Boot** LED on the MultiVOIP to go off before proceeding. This may take a couple of minutes.
5. At this time your VOIP network should be fully operational. Dial one of the sites in your network using the dialing directory supplied by your network Administrator.

MultiVOIP™ 100

Voice / Fax over IP Networks

Chapter 7

Using the MultiVOIP Software

MultiTech® Systems

Chapter 7 - Using the MultiVOIP Software

This chapter describes various features of the MultiVOIP software that enable you to change the configuration of your MultiVOIP. The default configuration parameters were entered during the loading of the software (Chapter 3). The MultiVOIP software and configuration utilities described in this chapter enable you to change that initial configuration as necessary.

The primary interface to the MultiVOIP software is a main menu (with **MultiVOIP 100 Setup** in the title bar) with individual buttons that enable you to quickly and easily select a desired function. These features are discussed in detail in the MultiVOIP Configuration section later in this chapter.

The other configuration utilities in the MVP120 software provide additional functionality.

Configuration port setup utility: Change the method by which you access the MultiVOIP (through a direct connection of a PC to the Command Port on the MultiVOIP, or via your Internet or LAN connection to the LAN port on the MultiVOIP).
Date and Time setup utility: Set the date and time used for data logging in the MultiVOIP.
Download Factory Defaults: Return the configuration to the original factory settings.
Download Firmware: Download new versions of firmware as enhancements become available.
Download User Defaults: Download your own default settings or profile.
Download Voice Coders: Download voice coders to the MultiVOIP.
Uninstall MultiVOIP Configuration utility: Remove the MVP120 software from your PC.
Upgrade Software utility: One step upgrade can be performed via serial port or over IP.

Before You Begin

The MultiVOIP software operates in a Microsoft Windows® environment. The MultiVOIP 100 program group contains icons for all the utilities described above. You can access the individual utility programs either by clicking **Start | Programs | MultiVOIP 100 | *(utility)***.

MultiVOIP Configuration

The MultiVOIP Setup menu enables you to display and change the voice channels and IP protocol parameters, display and manage the Phone Book listing, view statistics and call progress, and change features such as SNMP Agent, Telnet Server, WEB Server, and assign a MultiVOIP password.

The **Events** window in the lower third of the **Setup** menu provides information about the boot process.

The status bar at the bottom of the **Setup** menu displays the current status of the unit and shows, for example, if it is Running, the most recent date the unit was configured, the type of connection you have to the unit, and your rights. It shows if your PC is connected directly to the command port of the MultiVOIP or is communicating with the Ethernet port.

The **Rights** box displays whether the user has Read/Write or Read-Only rights. The first user to use the MultiVOIP has Read/Write rights and can view and change the configuration of the MultiVOIP. Subsequent users have only Read-Only rights and can view the configuration of the MultiVOIP but cannot change it.

Changing Channel Parameters

The channel parameters include the interface type and its options, voice and fax settings, and voice communications for the country and region in which the MultiVOIP is operating. The **Channel Setup** dialog box is accessed by clicking **Voice Channel** on the Setup menu.

InterfaceTab

The Interface tab defines the parameters related to the physical interface of the voice/fax channel.

The **Max Dial Digits** indicates the maximum number of digits the MultiVOIP will allow you to enter when dialing one of the numbers in the Phone Directory Database. As soon as you have entered this number of digits, the MultiVOIP will immediately attempt to match the digits you have dialed with an entry in the database. The range for the Max Dial Digits is from zero to 16 digits with a default of five.

The **Inter Digit Time** (in milliseconds) option in the **Dialing Options** group defines the amount of time the MultiVOIP waits between digits as they are entered by the user. If this timer expires, the MultiVOIP will immediately attempt to match the digits entered to an entry in the Phone Directory Database. The range for this option is 200 to 10,000 with a default of 2,000.

In the **Flash Hook Time** box, enter the time, in milliseconds, for the duration of flash hook signals output on the FXO interface.

The **Auto Call** option allows the local MultiVOIP to call a remote MultiVOIP without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote MultiVOIP that you identified in the Remote MultiVOIP Phone Number box of this option.

Check with your in-house telephone personnel to verify whether your local PBX dial signalling is **Pulse** or **DTMF** (tone). Select the regeneration option accordingly.

For FXO-to-FXO communications, you can enable a specific kind of FXO disconnect: **Current Loss**, **Tone Detection**, or **Silence Detection**. Check with your in-house phone personnel to verify the type of disconnect to use. If **Current Loss** is checked, the VOIP will hang up when it detects a loss of current on the FXO port. For tone detection, select from the lists one or two tones that will cause the line to disconnect. The person hanging up the call must then press the key or keys that produce those tones. For silence detection, select **One Way** or **Two Way**, then set the timer for the number of seconds of silence before disconnect. The default value of 15 seconds may be shorter than desired for your application.

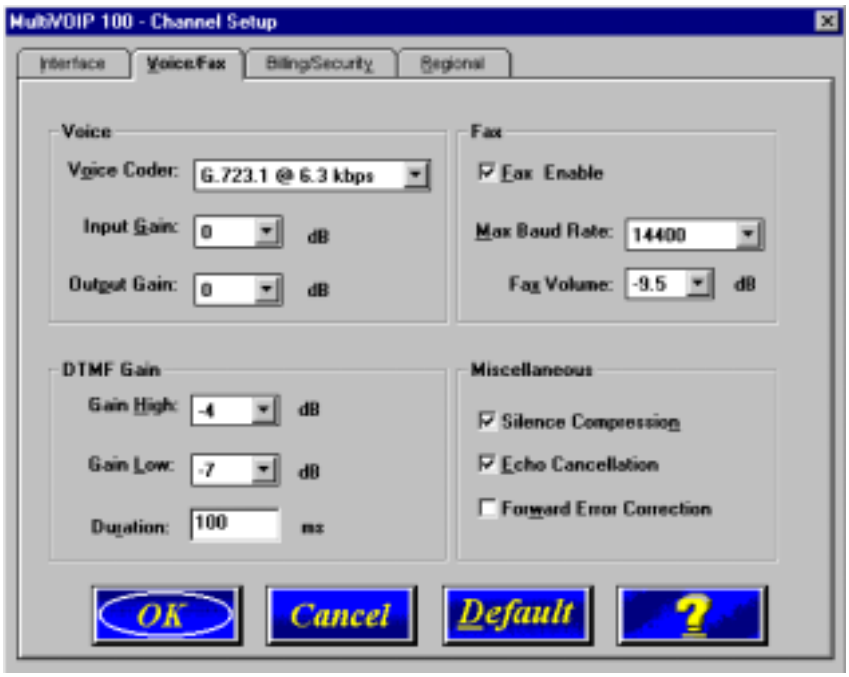
To dedicate the local voice/fax channel to a remote voice/fax channel, (so you will not have to dial the remote channel) select the **Auto Call Enable** check box. Enter the phone number of the remote VOIP in the **Phone Number** box.

The **Voice/Fax** tab displays the parameters for the voice coder, faxing, and DTMF gain.

The **FXO Options** group controls how the selected channel interacts with answering machines. If the **Current Loss** check box is selected, the local MultiVOIP interrupts power from the Phone port when a call is hung up on the remote end. The remote MultiVOIPs must also be configured for current loss.

Voice/Fax Tab

The **Voice/Fax** tab controls the Voice Coder, Input and Output gains, Fax settings, DTMF gain, and some miscellaneous options.



The MultiVOIP supports many state-of-the-art ITU (International Telecommunications Union) voice coders. The **Voice Coder** list enables you to select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling has to have the same voice coder selected. Otherwise, you will always get a Fast Busy signal.

The **Input Gain** and **Output Gain** lists enable you to select the amount of gain you want. Gain is the increased signalling power that occurs as the signal is boosted by the MVP120. The **Input Gain** list defines the voice input gain for the Voice/Fax channel. Before the MVP120 digitizes voice, the volume can be increased or decreased. Input gain modifies the level of audio coming into the voice channel before it is sent over the Internet to a remote MultiVOIP. The valid range for this option is +31dB to -31 dB. The recommended and default value is **0**. The **Output Gain** list defines the voice output gain for the Voice/Fax channel. Before the MVP120 converts digital voice back to analog, the volume can be increased or decreased. The output gain modifies the level of the audio being output to the device attached to the Voice/Fax channel. The valid range for this option is +31dB to -31 dB. The recommended and default value is **0**.

The **Fax** group enables a fax machine to transmit and receive faxes through the MultiVOIP. If a fax machine is connected to the voice/fax channel, the **Max Baud Rate** should be set to match the baud rate of the fax machine. Refer to user documentation for your fax. The **Fax Volume** setting controls the output level of the fax tones, and this setting should be changed only under the direction of Multi-Tech's Technical Support personnel.

Miscellaneous Group

The **Silence Compression** check box defines whether silence compression is enabled for the selected voice channel. If enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth being used by the voice/fax channel.

The **Echo Cancellation** check box defines whether echo cancellation is enabled for the voice/fax channel. If enabled, the MultiVOIP will remove echo-delay, which improves sound quality.

The **Forward Error Correction (FEC)** check box enables recovery of some corrupted or lost voice packets. FEC adds 50 % overhead to the total network bandwidth.

The **DTMF Gain** group controls the volume level of the digital tones sent out for Touchtone dialing. The **Gain High** and **Gain Low** boxes control the gain in dB (decibels) of the High and Low tones in the tone pairs; the default gain values are -4 dB and -7 dB, respectively. DTMF Gain should not be changed except under supervision of Multi Tech's Technical Support.

Billing/Security Tab

The **Billing/Security** tab displays the parameters for billing options, call authentication, and

automatic disconnection.

You can set up billing options for inbound and outbound calls by selecting them in the **Billing Options** group and then entering the charge in cents per number of seconds.

The **Call Authentication** option enables password protection for outbound and inbound calls on the selected voice/fax channel. If you enable password protection on inbound/outbound calls, you need to enter a password of up to 14 numeric characters in the **Password** box.

The **Automatic Disconnect** option limits call duration to the number of seconds entered in the **Disconnect After** box. The default value of 180 seconds can be changed to any other value up to 65,535 (roughly 18.2 hours).

Regional Tab

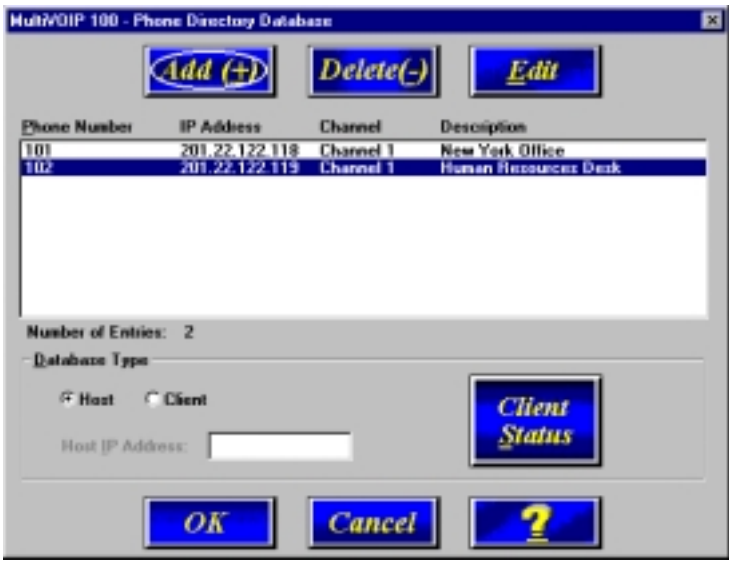
The regional tab controls the voice communications for the country or region in which the MultiVOIP is being used.

From the **Country/Region** list you can select the country or region for which you are configuring the MultiVOIP. The **Tone Pairs** group always displays the tones used in the country or region currently selected. In addition to Australia, Central America, Chile, Europe, France, Japan, UK, and USA, there is a Custom selection (with defaults identical to USA) that will make the **Custom** button active. Clicking **Custom** enables you to edit the Tone Pairs and establish custom sets of tone pairs for Dial Tone, Ring, and Busy on a **Custom Tone Pair Settings** dialog box.

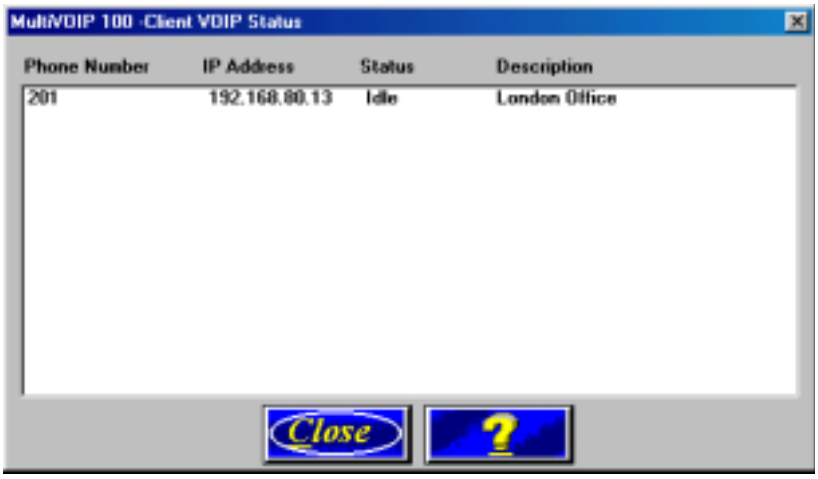
The **Pulse Generation Ratio** group contains two ratios: the 60/40 is for the USA, and the 67/33 ratio is for international applications.

Changing the Phone Directory Database

The **Phone Directory Database** dialog box displays all the phone numbers in your MultiVOIP network. The database displays the phone numbers in numerical order with the IP Address, Channel assignment, and description.



Access this database by clicking **Phone Book** on the MultiVOIP Main menu. You can add, delete, or edit any entry in the database and you can change the host-client relationship of the database. Click **Client Status** to display the status of all the client units in your VOIP network. The Phone Number of each client displays with its IP Address, current line status, and the description of the phone number.



The phone number does not have to be a conventional phone number such as 555-1234. It can be a single digit or several digits, except it cannot be longer than the entry in the **Max Dial Digits** box in the **Channel Setup** dialog box. For example, you could enter a phone number of 101 with a description of New York Office. If you want the call to be rolled over to a second channel, select the **Permit Hunting** check box. If the assigned channel is busy, then the call is rolled over to the next channel.

Note: Proprietary firmware does not allow hunting across different MultiVOIP channels. It does allow hunting between channels on the same MultiVOIP.

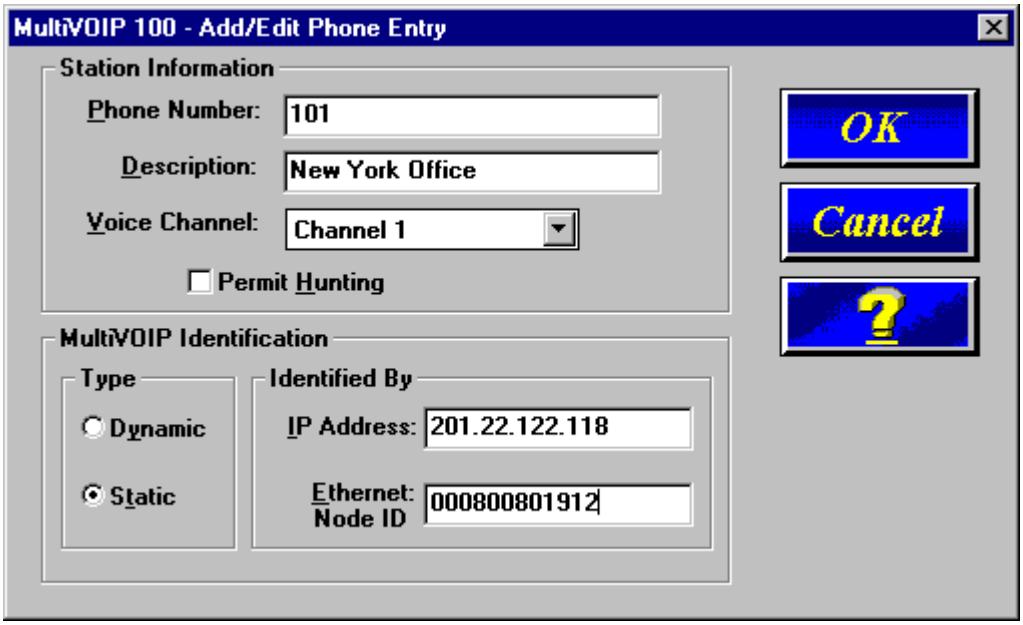
The **MultiVOIP Identification** group defines the type of addressing (**Dynamic** or **Static**) for the host and client units for their respective Phone Numbers. The Phone Numbers assigned to the host MultiVOIP have to be Static addressing and the Phone Numbers assigned to the client MultiVOIP can be either dynamic or static depending on whether a Proxy Server is providing the connection to the Internet. If a Proxy Server is in front of the MultiVOIP providing the Internet connection and the Proxy Server is using dynamic addressing (the ISP is assigning the Proxy Server IP address), then this client MultiVOIP will be defined as using Dynamic addressing and the **IP Address** box in the **Identified By** group will be unavailable.

If a Proxy Server with a static IP address is in front of the client MultiVOIP, then the **Identified By** IP Address box must contain the public or WAN IP address of the Proxy Server.

If the client MultiVOIP is connected directly to the Internet, then its addressing mode must be Static. If the client unit is using Static addressing, then the **IP Address** box in the Identified By group has to contain the Static IP address of the client MultiVOIP.

The **Ethernet Node ID** is a 12-digit Identification Number assigned to each unit. This Ethernet Node ID number is a hardware identification number that is affixed to each unit during the manufacturing process and cannot be changed. This ID number (for example, 0008005xxxxx) is located on an ID label attached to every unit. This ID number has to be entered in the **Ethernet Node ID** box for the telephone number entered in the **Phone Number** box. If you are assigning a Phone Number for a client unit, the Ethernet Node ID has to be for that client unit.

If this label is damaged or missing, you can also obtain the ID number by Telneting to the unit. From the MultiVOIP Telnet Server menu, choose the Voice over IP Configuration option which takes you to the Main Menu. In the Main Menu, choose System Information and the ID number is presented in the Ethernet Port Address of the System Information menu.



When you enter this information and click **OK**, the information is loaded into the phone directory database.

To add a second entry, click **Add** and the **Add/Edit Phone Entry** dialog box displays again. After the host entries are added to the database, the same data has to be added for each client MultiVOIP.

To establish the phone directory database for a client MultiVOIP, you do not have to enter phone numbers, but you have to check the **Client** option in the **Database Type** group. When you click the Client option, the Update Database From group becomes active. Enter the IP address of the Host MultiVOIP. Once online, the client MultiVOIP will receive the phone directory database from the Host.

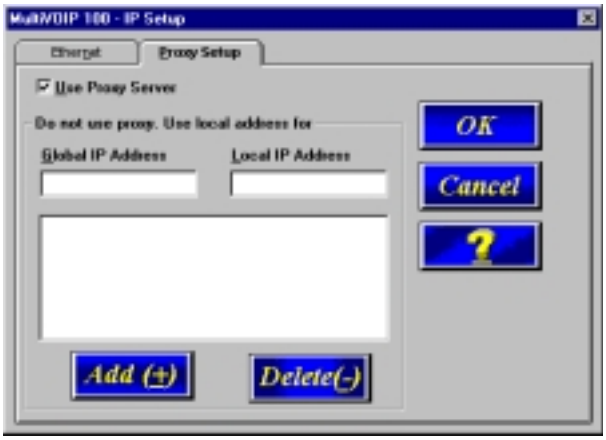
Changing IP Parameters

The IP Setup dialog box establishes the IP addressing for the local Ethernet LAN, defines the Internet gateway address, and for a client MultiVOIP the global-to-local IP address translation is defined on the **Proxy Setup** tab. The **IP Setup** dialog box is accessed by clicking **IP** on the MultiVOIP 100 Setup menu.

When the **IP Setup** dialog box displays, the IP address of your MultiVOIP displays with its IP Mask. The Gateway Address is the IP address of the device connected to the Internet.

Proxy Server

The **Proxy Setup** tab is used when a Proxy Server is used to connect the LAN to the Internet.



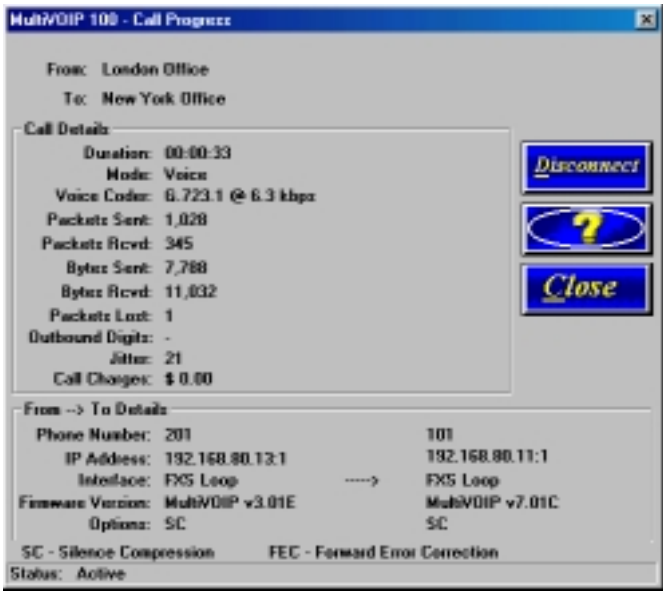
If a Proxy Server is used in a MultiVOIP network, an address translation takes place within the Proxy Server to direct the phone call to the correct MultiVOIP.

When a MultiVOIP's connection to the Internet is through a Proxy Server, the WAN port on the Proxy Server must have a static registered IP address. Remote MultiVOIPs will only be able to access a MultiVOIP located behind a Proxy Server at the static IP address. This static IP address will be used in the Phone Directory Database when assigning directory numbers to this MultiVOIP.

The **Global IP Address** box must contain the static IP address of the WAN port of the Proxy Server. The **Local IP Address** box must contain the local IP address of the MultiVOIP. In this case the local IP address is not used in the Phone Directory Database. There must be a unique static IP address on the Wan side of the proxy server for each MultiVOIP located behind the proxy server.

Viewing Call Progress

The **Call Progress** dialog box displays the status of a call in progress. This dialog box is accessed from the **Setup** menu by clicking **Call Progress**.



The ratio of Packets Lost versus Packets Received provides a general indication of the integrity of the Internet connection. To reduce the frequency of lost packets, select a low-bit-rate coder, such as G.723 or Netcoder. Selecting the **Forward Error Correction** check box on the **Voice/Fax** tab in the **Channel Setup** dialog box enables the MultiVOIP to recover many of the lost packets.

The Jitter (ms) value indicates the mean deviation of the difference in packet spacing at the receiver compared to the sender for a pair of packets.

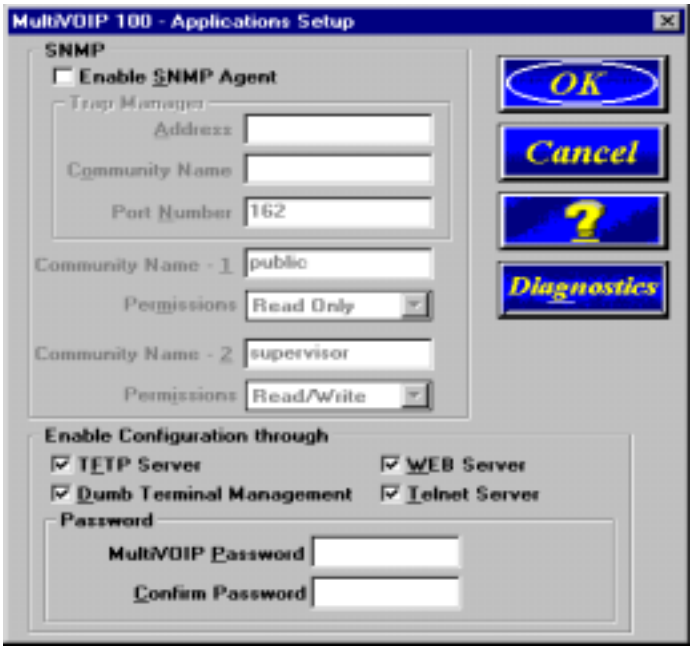
Applications Setup

Clicking **Others** on the Setup menu displays the **Applications Setup** dialog box. You can select the **Enable SNMP Agent** check box and set up all the necessary parameters, enable or disable various remote configuration methods such as TFTP (Trivial File Transfer Protocol) Server, Web Server, Dumb Terminal Management, and Telnet Server; and assign a Password to the MultiVOIP for Internet security. These applications enable remote viewing and changing of the MultiVOIP configuration from anywhere on the connected internetwork.

Verify that the desired applications are selected. The default condition is all applications are checked. To disable a given application, click to uncheck the check box and disable support.

SNMP related operations can be performed only when the **SNMP Agent** is enabled on this dialog box. The IP address of the system, such as SNMP Manager that will receive the Traps from the MultiVOIP should be entered in the **IP Address** box. The **Community Name** of the SNMP Manager receiving the Traps can be a maximum of 19 characters and is case sensitive. The default **Port Number** of the SNMP Manager receiving the Traps is 162. The MultiVOIP currently supports a maximum of two community users at a time, and they can be assigned either Read/Write or Read-Only rights.

The **Password** group enables you to enter a password, up to 13 alphanumeric characters, to be used for Internet Security. Once the password is entered in the **MultiVOIP Password** box and confirmed in the **Confirm Password** box, users will be queried to enter the password before gaining access to the MultiVOIP. The same password is used for all configuration methods.



For more information on using these applications, refer to Chapter 8, Remote Configuration.

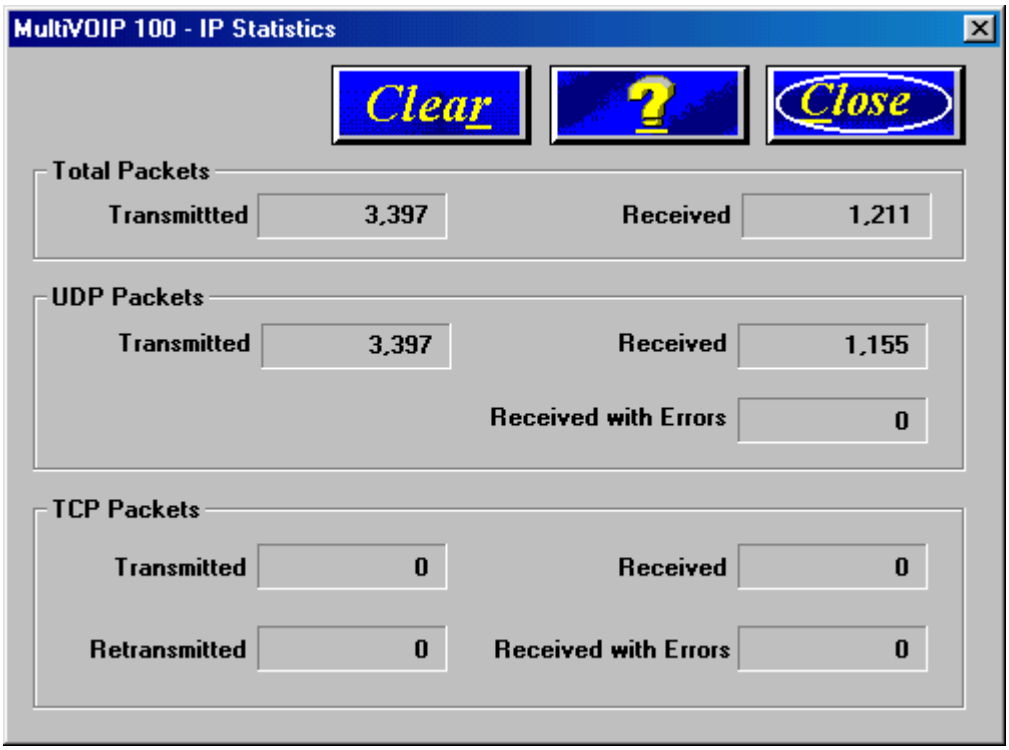
Viewing Statistics

The Statistics dialog box enables you to view statistics for major events of the MultiVOIP operation. This dialog box is accessed by clicking **Statistics** on the **Setup** menu.

Statistics can be a helpful troubleshooting tool. For example, viewing the Voice Channel statistics you can see the attempted and completed calls, call duration, average call length, bytes/packets sent and received, and so on.

IP Statistics

IP is a connection-less network protocol residing in the network layer of a conventional OSI layered model (for more information on this model, refer to Appendix A). Depending on what is going on at the application layer, IP will typically use one of two transport layer protocols: User Datagram Protocol (UDP), a connection-less transport layer protocol used with TFTP or SNMP; and Transport Control Protocol (TCP) is a connection-oriented transport layer protocol used with FTP, Telnet, and SNMP.



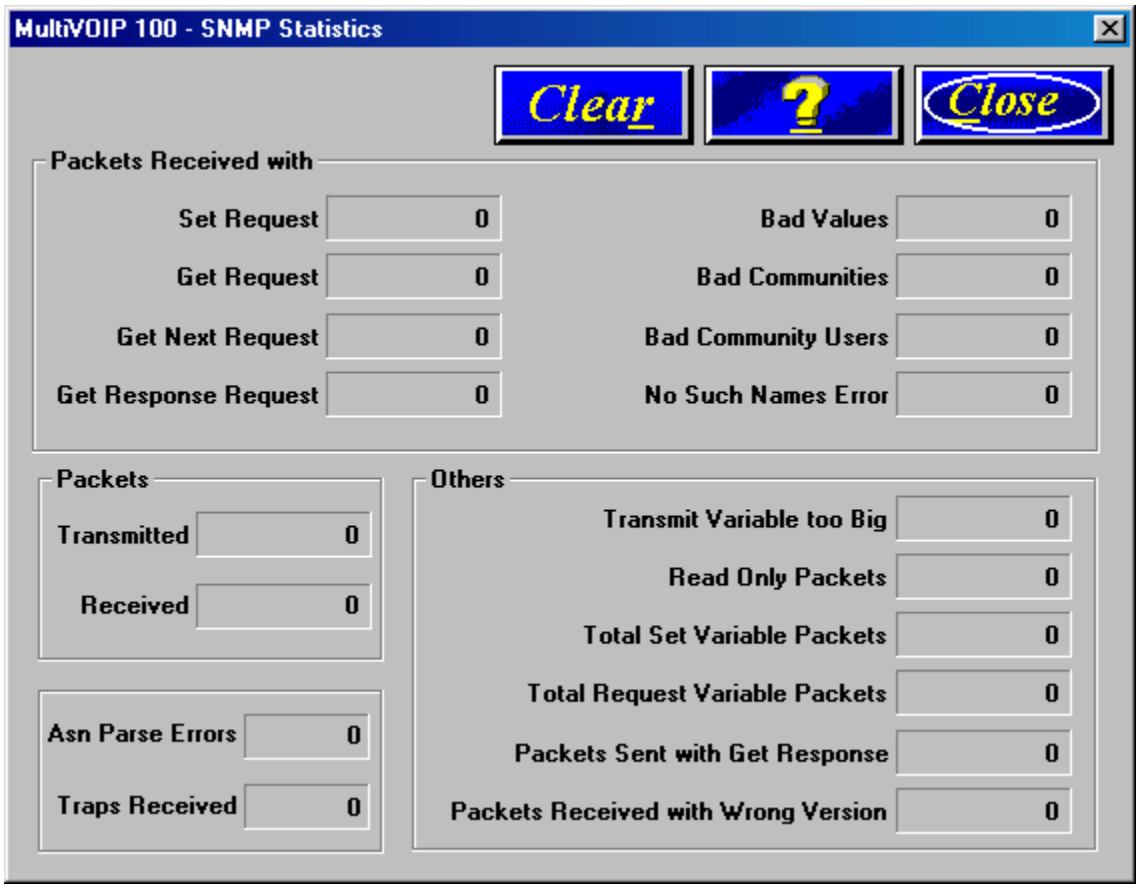
UDP makes use of the port concept and has no measures for flow control, reliability, or error recovery. It is used when the full services of TCP are not required, and the reliability measures must be assumed by another layer.

TCP works well in environments where the reliability measures are not assumed by other layers. It is connection-oriented and has a full range of services.

For the most part these statistics are informational, and their use as a troubleshooting tool will be contingent on the applications running in the upper layers. For example, if you are having problems connecting to the MultiVOIP's Web server, you would look under the TCP section to see if any packets are being transferred. If not, that may indicate the Web server is not enabled. Or, if you are having problems establishing a remote connection through TFTP, you could look in the UDP section to see if any packets are being received. If not, you may need to review your network addressing.

SNMP Statistics

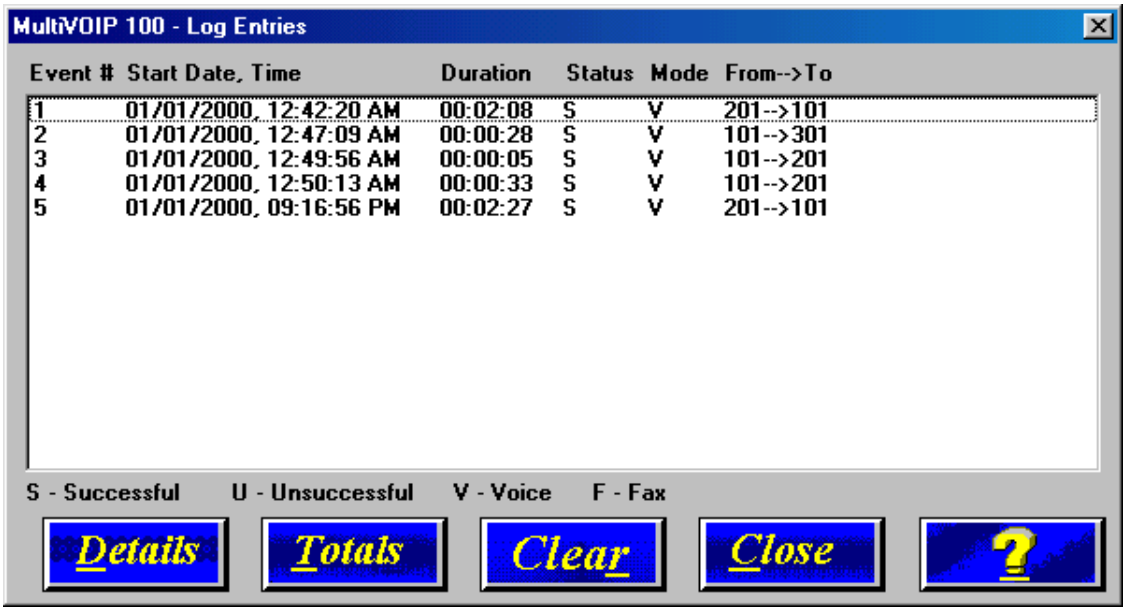
The SNMP Statistics dialog box provides statistical information on Simple Network Management Protocol (SNMP).



SNMP is an application layer protocol that facilitates the exchange of management information between network devices. There are three key components in SNMP; the devices that are to be managed, agents, and the network management systems. The managed device is the network device, like a router. The agent is the software module residing in the managed device pertaining to network management. The network management systems runs the SNMP application that controls the managed devices and monitors theirs status. Four primary operations (Set, Get, Get Next, and Trap) are performed using SNMP.

Viewing Logs

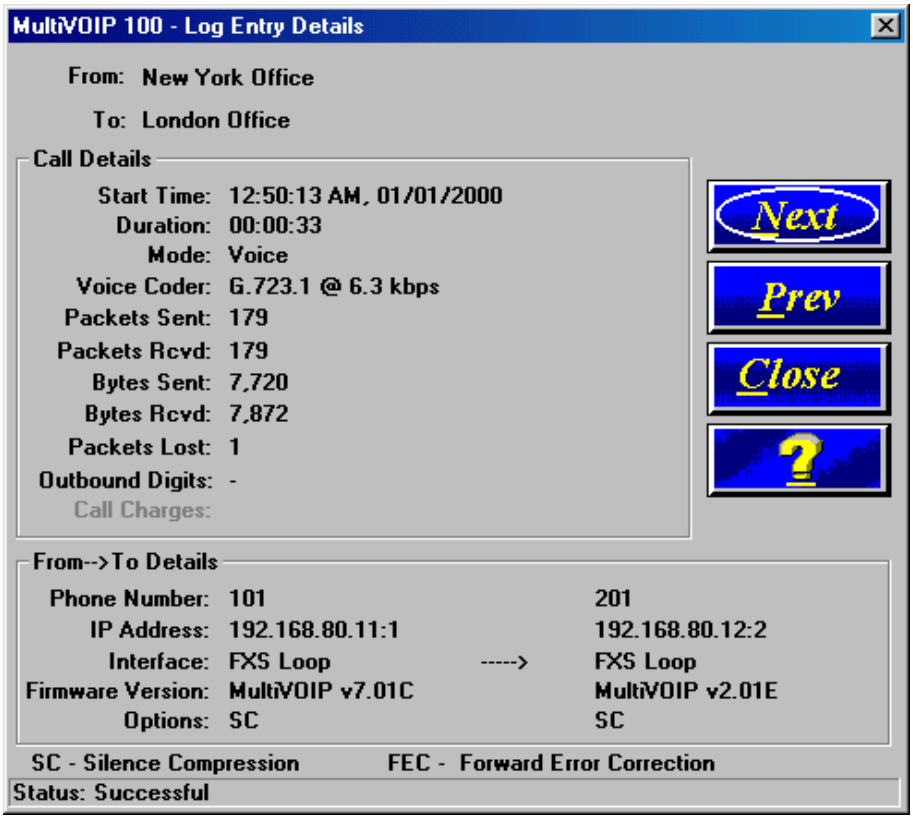
The **Log Entries** dialog box displays a chronological history of all calls into and out of this unit. This dialog box is accessed by clicking **Logs** in the **Statistics** dialog box.



The **Log Entries** dialog box displays each call as a sequentially numbered Event with the date, time, duration of the call, the status of the call (**S**uccessful or **U**nsuccessful), Mode (**V**oice or **F**ax), and the from and to numbers.

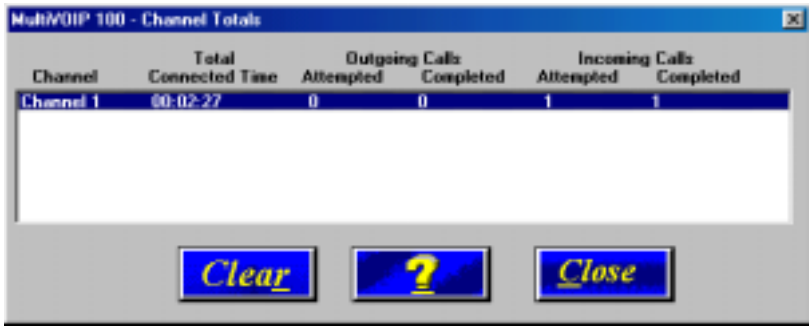
Viewing Log Entry Details

The **Log Entry Details** dialog box displays the status of a completed call. This dialog box displays the same details as the **Call Progress** dialog box after a call is completed.



Viewing Channel Totals

The **Channel Totals** dialog box displays Outgoing and Incoming calls with their Attempted and Completed numbers for each channel on this MultiVOIP. The Total Connected Time for the channel also displays.



Reports

A report of the contents of the **Log Entries** dialog box can be generated using the Windows Notepad accessory and then printed from your local PC. The report is generated by entering the To and From dates in the **Report Generation** dialog box and then clicking **Generate**. This function provides a hard copy of the **Log Entries** dialog box.

Upgrade Procedures

Whenever you upgrade your version of the MultiVOIP software, you must first install the new software on your PC. Then, download the Firmware, the Factory Defaults, and the Voice Coders to upgrade the MultiVOIP.

Before starting the upgrade process, view the current configuration and write down important data such as your IP address, phone book contents, and voice channel configurations. These settings must be put back in place after the software has been upgraded.

Two utility programs included in the MultiVOIP software are to be used only after the unit has been repaired or upgraded. They are Download Firmware and Download Voice Coders.

You can also use the **Upgrade Software** utility. This is recommended. This utility upgrades the boot code, factory defaults, firmware, and voice coders in one step. After the files download to the MultiVOIP, the unit reboots and is ready for use with upgraded firmware and voice coders.

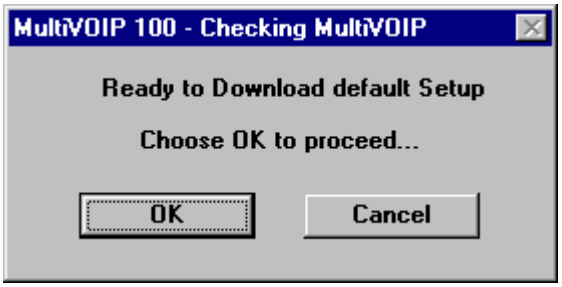
One-Step Upgrade

Click **Start** | **Programs** | **MultiVOIP 100 v701C** | **Upgrade Software**.

Click **Yes**. The download process begins.

Enter your LAN IP Address, Subnet Mask, and Gateway Address in the **IP Protocol Default Setup** dialog box. Click **OK** when finished.

Enter your current configuration in the **Channel Setup** dialog box. Click **OK** when finished.



Click **OK**. The MultiVOIP firmware and coders download, then the MultiVOIP reboots.

Manual Upgrade Procedure

Use this procedure only if you want to manually upgrade your MultiVOIP.

Note: Steps 4-8 may be performed via the command port or over IP.

1. Run MultiVOIP Configuration from your old version of the MultiVOIP software. Note the current settings. Your MultiVOIP will be reset to factory defaults during this upgrade.
2. Unistall your old version of MultiVOIP software by selecting the **Unistall MultiVOIP Configuration** option from the program group.
3. Install the proprietary software from the MultiVOIP CD.
4. a. From the program group of the MultiVOIP software, select **Download Firmware**.
   b. Enter "***.upg**" in the **File name** box and press ENTER.
   c. Select the file named **boot_100.upg**, if you are using the command port.
   OR
   Select **tftp.upg** if you are using IP.
5. From the program group of the MultiVOIP software, select **Download Factory Defaults**.
6. From the program group of the MultiVOIP software, select **Download Firmware**. Select the file named **mtvoip.bin**.
7. From the program group of the MultiVOIP software, select **Download Voice Coders**. Select the default file.
8. Select MultiVOIP Configuration from the program group and reconfigure your MultiVOIP. The upgrade is complete.

MultiVOIP™ 100

Voice / Fax over IP Networks

Chapter 8

Remote Configuration

MultiTech® Systems

Chapter 8 - Remote Configuration and Management

This chapter provides procedures for viewing or changing the configuration of a remote unit. Two methods are provided to access a remote unit; the first method is modem based and the second method is using IP. Within the IP method, three applications can be used: 1) LAN-Based using TFTP (Trivial l File Transfer Protocol), 2) Telnet as a client application, or 3) a standard web browser on the Internet.

Remote configuration requires the MultiVOIP software to be loaded on the local PC. The local PC then controls the remote MultiVOIP using either the modem connection or the Ethernet link.

Modem-Based

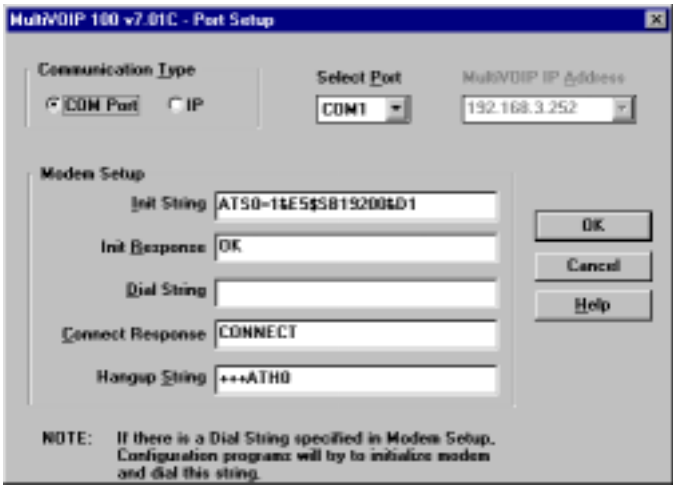
To remotely configure a MultiVOIP, a local PC needs to be connected to a dial-up line and the MultiVOIP software configured to call the remote MultiVOIP. The remote MultiVOIP needs to have a modem connected to a dial-up line and the Command Port. Once the connection to the remote unit is made, you can change the configuration as you see fit. Once the configuration is changed, you can down load the new configuration to the remote MultiVOIP. Refer to the Modem-Based Remote Configuration Procedure in this chapter to remotely configure a MultiVOIP.

1. At the remote site, remove the serial cable from the PC to the Command Port connector on the back panel of the MultiVOIP.
2. At the remote site, connect a special cable (Remote Configuration Cable) from the Command Port connector on the back panel of the MultiVOIP to the RS232 connector on the modem. The special cable is a serial cable with male connectors on both ends. Refer to Appendix B for cable details.

   Connect the modem to your local telephone line.

   Provide your telephone number to the person verifying your configuration.

   Configure the modem serial port for 19200 bps and turn on Force DTR.
3. At the main site, connect your local PC to a modem that is connected to a dial-up line.
4. Install the MultiVOIP software on the local PC. When installed, click **Start | Programs | MultiVOIP 100 | Configuration Port Setup**, or double click on the **Configuration Port** icon in the MultiVOIP 100 program group.
5. The **MultiVOIP 100 Setup** dialog box displays.



Verify that the **Communication Type** is set for **COM Port** and the **Select Port** box is set for the COM port of your local PC.

In the **Dial String** box, enter the AT command for dialing (ATDT) plus the phone number of the remote MultiVOIP.

If your Modem Initialization String, Initialization Response, or Connect Response values are different than the defaults in the dialog box, refer to your modem user documentation and change the default values to match your modem.

Click **OK** when you are satisfied with your selections.

6. Run the MultiVOIP Configuration program. Click **Start | Programs | MultiVOIP 100 | MultiVOIP Configuration**, or double-click the **MultiVOIP Configuration** icon in the program group.

7. The **Dialing** dialog box displays while software is dialing the remote modem.

8. The **Reading Setup** dialog box displays.

9. The **MultiVOIP 100 Setup** menu displays. This is the dialog box of the remote MultiVOIP. Refer to the online Help provided with your software for a description of each dialog box and box within a dialog box.

10. After you have changed the configuration of the remote MultiVOIP, click **Download Setup** to update the configuration. The remote MultiVOIP will be brought down, the new configuration written to the unit, and the unit will reboot.

11. Click **Exit** after the download completes.

12. The **Hangup connection?** dialog box displays. Click **Yes** to disconnect the phone connection to the remote site.

13. If the same telephone number is not going to be used again in the immediate future, you can remove it from the **Port Setup** dialog box.

LAN-Based

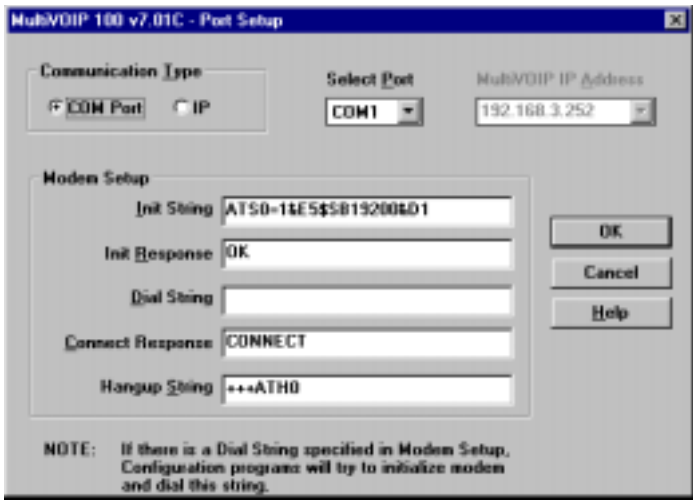
The LAN-based remote configuration requires a Windows Sockets compliant TCP/IP stack. TCP/IP protocol software must be installed and functional before the configuration program can be used.

1. You must assign an Internet (IP) address for the PC and for each node that will be managed by the configuration program. Refer to the protocol software documentation for instructions on how to set the IP addresses.

   Once you have completed this step, you should be able to use the protocol Ping command for the PC host name. You should also test the network interface configuration by Pinging another TCP/IP device that is connected to the network.

2. Install the MultiVOIP software on the local PC. When installed click **Start | Programs | MultiVOIP 100 | Configuration Port Setup**, or double-click on the **Configuration Port Setup** icon in the **MultiVOIP 100** program group.

3. The **MultiVOIP Port Setup** dialog box displays.



   Verify that the **IP** option is selected in the **Communication Type** group.

   In the **MultiVOIP IP Address** box, enter the IP Address of the remote MultiVOIP.

4. Click **OK** when you are satisfied with your selections.

5. Run the MultiVOIP Configuration program. Click **Start | Programs | MultiVOIP 100 | MultiVOIP Configuration**, or double-click on the **MultiVOIP Configuration** icon in the **MultiVOIP 100** program group.

6. The **MultiVOIP 100 Setup** dialog box displays. This is the dialog box of the remote MultiVOIP. Refer to the online Help provided with your MultiVOIP for the definition of each dialog box.

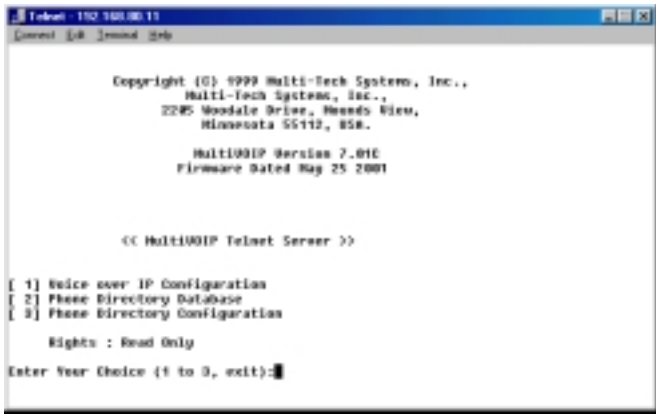
7. After you have changed the configuration of the remote MultiVOIP, click **Download Setup** to update the configuration. The remote MultiVOIP will be brought down, the new configuration written to the unit, and the unit will reboot.
8. Click **Exit** when the downloading is complete.
9. Double-click the **MultiVOIP Configuration** icon in the program group to verify that the MultiVOIP is running.

Telnet

A typical Telnet client application is described next. The MultiVOIP has a built-in Telnet Server that enables Telnet client PCs to access the MultiVOIP. A typical Telnet client is allowed to configure the MultiVOIP. In addition, the MultiVOIP can be remotely accessed and configured from any where on the Internet through its Web interface.

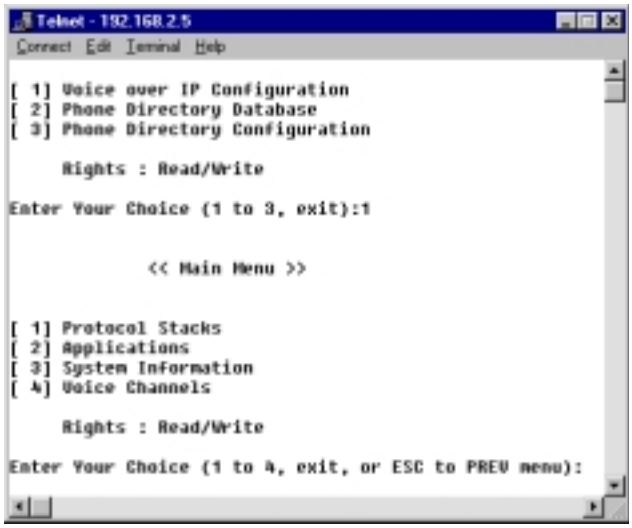
MultiVOIP Telnet Server Menu

The MultiVOIP Telnet Server menu provides three basic options: Voice over IP Configuration, Phone Directory Database, and Phone Directory Configuration. A further option enables you to close the Telnet session.


Copyright (C) 1999 Multi-Tech Systems, Inc.,
Multi-Tech Systems, Inc.,
2205 Woodale Drive, Mounds View,
Minnesota 55112, USA.
MultiVOIP Version 7.01E
Firmware Dated May 25 2001
<< MultiVOIP Telnet Server >>
[1] Voice over IP Configuration
[2] Phone Directory Database
[3] Phone Directory Configuration
Rights : Read Only
Enter Your Choice (1 to 3, exit):

Voice Over IP Configuration

Selecting **Option 1** displays the main menu, which allows further configuration options. These options include Protocol Stacks (option 1), Applications (option 2), System Information (option 3), and Voice Channels (option 4).


Telnet - 192.168.2.5
Connect Edit Terminal Help
[1] Voice over IP Configuration
[2] Phone Directory Database
[3] Phone Directory Configuration
Rights : Read/Write
Enter Your Choice (1 to 3, exit):1
<< Main Menu >>
[1] Protocol Stacks
[2] Applications
[3] System Information
[4] Voice Channels
Rights : Read/Write
Enter Your Choice (1 to 4, exit, or ESC to PREV menu):

Phone Directory Database

Selecting **Option 2** enables you to add entries to the **Phone Directory Database**. Refer to Chapter 6, Using the MultiVOIP Software, for more details on the database.

Phone Directory Configuration

Selecting **Option 3** enables you to configure and manage the **Phone Directory**. The various options are described in detail in Chapter 6, Using the MultiVOIP Software.

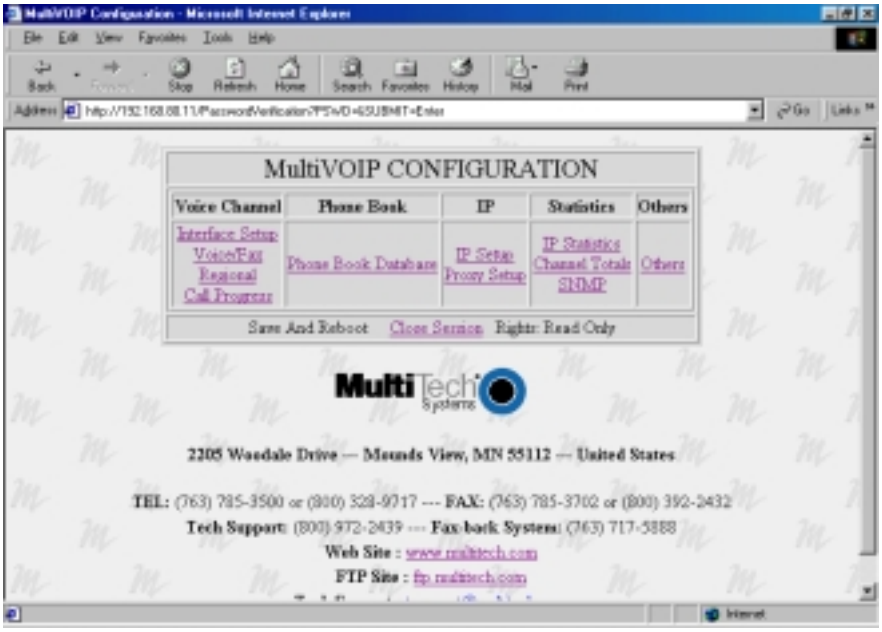
Web Management

The MultiVOIP can be accessed, using a standard Web browser, from anywhere on the connected Internet. In order to provide this support, select the **WEB Server** check box in the **Applications Setup** dialog box (see Chapter 6, Using the MultiVOIP Software).

Once enabled, users can access the MultiVOIP by entering its IP address in the **Address** box of their Web browser. The following Web page appears.

Welcome to MultiTech's MultiVOIP Configuration
MultiVOIP Version 7.01C
Firmware Dated May 25 2001
Password :
Enter
2205 Woodale Drive — Mounds View, MN 55112 — United States
TEL: (763) 785-3500 or (800) 328-9717 --- FAX: (763) 785-3702 or (800) 392-2432
Tech Support: (800) 972-2439 --- Fax-back System: (763) 717-5888
Web Site : www.multitech.com

If a Password was entered in the **Applications Setup** dialog box, enter the password and click **Enter**.



From this Web page, you can access all the configuration options. Refer to Chapter 7, Using the MultiVOIP Software, for a description of the various options.

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Chapter 9

Warranty, Service,and Technical Support

MultiTech® Systems

Chapter 9 - Warranty, Service, and Technical Support

This chapter starts out with statements about your MultiVOIP warranty. The next section, Technical Support, should be read carefully if you have questions or problems with your MultiVOIP. It includes the technical support phone numbers, space for recording your product information, and an explanation of how to send in your MultiVOIP should you require service. The final section explains how to get support through the Internet.

Multi-Tech Systems, Inc. Warranty & Repairs Policies

Warranty

Multi-Tech Systems, Inc., (hereafter “MTS”) warrants that its products will be free from defects in material or workmanship for a period of two, five, or ten years (depending on model) from date of purchase, or if proof of purchase is not provided, two, five, or ten years (depending on model) from date of shipment.

MTS MAKES NO OTHER WARRANTY, EXPRESS OR IMPLIED, AND ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE HEREBY DISCLAIMED.

This warranty does not apply to any products which have been damaged by lightning storms, water, or power surges or which have been neglected, altered, abused, used for a purpose other than the one for which they were manufactured, repaired by Customer or any party without MTS’s written authorization, or used in any manner inconsistent with MTS’s instructions.

MTS’s entire obligation under this warranty shall be limited (at MTS’s option) to repair or replacement of any products which prove to be defective within the warranty period or, at MTS’s option, issuance of a refund of the purchase price. Defective products must be returned by Customer to MTS’s factory – transportation prepaid.

MTS WILL NOT BE LIABLE FOR CONSEQUENTIAL DAMAGES, AND UNDER NO CIRCUMSTANCES WILL ITS LIABILITY EXCEED THE PRICE FOR DEFECTIVE PRODUCTS.

Repair Procedures for U.S. and Canadian Customers

In the event that service is required, products may be shipped, freight prepaid, to our Mounds View, Minnesota factory:

Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, MN 55112
Attn: Repairs, Serial # _____

A Returned Materials Authorization (RMA) is not required. Return shipping charges (surface) will be paid by MTS.

Please include, inside the shipping box, a description of the problem, a return shipping address (must have street address, not P.O. Box), your telephone number, and if the product is out of warranty, a check or purchase order for repair charges.

For out of warranty repair charges, go to www.multitech.com/documents/warranties
Extended two-year overnight replacement service agreements are available for selected products. Please call MTS at (888) 288-5470, extension 5308 or visit our web site at http://www.multitech.com/programs/orc/ for details on rates and coverage’s.

Please direct your questions regarding technical matters, product configuration, verification that the product is defective, etc., to our Technical Support department at (800) 972-2439 or email tsupport@multitech.com. Please direct your questions regarding repair expediting, receiving, shipping, billing, etc., to our Repair Accounting department at (800) 328-9717 or (763) 717-5631, or email mtsrepair@multitech.com.

Repairs for damages caused by lightning storms, water, power surges, incorrect installation, physical abuse, or user-caused damages are billed on a time-plus-materials basis.

Repair Procedures for International Customers (Outside U.S.A. and Canada)

Your original point of purchase Reseller may offer the quickest and most economical repair option for your Multi-Tech product. You may also contact any Multi-Tech sales office for information about the nearest distributor or other repair service for your Multi-Tech product. http://www.multitech.com/COMPANY/offices/DEFAULT.ASP

In the event that factory service is required, products may be shipped, freight prepaid to our Mounds View, Minnesota factory. Recommended international shipment methods are via Federal Express, UPS or DHL courier services, or by airmail parcel post; shipments made by any other method will be refused. A Returned Materials Authorization (RMA) is required for products shipped from outside the U.S.A. and Canada. Please contact us for return authorization and shipping instructions on any International shipments to the U.S.A. Please include, inside the shipping box, a description of the problem, a return shipping address (must have street address, not P.O. Box), your telephone number, and if the product is out of warranty, a check drawn on a U.S. bank or your company's purchase order for repair charges. Repaired units shall be shipped freight collect, unless other arrangements are made in advance.

Please direct your questions regarding technical matters, product configuration, verification that the product is defective, etc., to our Technical Support department nearest you or email tsupport@multitech.com. When calling the U.S., please direct your questions regarding repair expediting, receiving, shipping, billing, etc., to our Repair Accounting department at +(763) 717-5631 in the U.S.A., or email mtsrepair@multitech.com.

Repairs for damages caused by lightning storms, water, power surges, incorrect installation, physical abuse, or user-caused damages are billed on a time-plus-materials basis.

Repair Procedures for International Distributors

Procedures for International Distributors of Multi-Tech products are on the distributor web site.

http://www.multitech.com/PARTNERS/login/

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10-Sep-01

Online Warranty Registration

To register your MultiVOIP online, click the following link:

http://www.multitech.com/register/

Service

If your tech support specialist decides that service is required, your MultiVOIP may be sent (freight prepaid) to our factory. Return shipping charges will be paid by Multi-Tech Systems.

Include the following with your MultiVOIP:

- a description of the problem
- return billing and return shipping addresses
- contact name and phone number
- check or purchase order number for payment if the MultiVOIP is out of warranty. Check with your technical support specialist for the standard repair charge for your MultiVOIP.
- if possible, note the name of the technical support specialist with whom you spoke.

If you need to inquire about the status of the returned product, be prepared to provide the serial number of the product sent.

Send your MultiVOIP to this address:

MULTI-TECH SYSTEMS, INC.
2205 WOODALE DRIVE
MOUNDS VIEW, MINNESOTA 55112
ATTN: SERVICE OR REPAIRS

You should also check with the supplier of your MultiVOIP on the availability of local service and loaner units in your part of the country.

Technical Support

Multi-Tech has an excellent staff of technical support personnel available to help you get the most out of your Multi-Tech product. If you have any questions about the operation of this unit, call 1-800-972-2439. Please fill out the MultiVOIP information (below), and have it available when you call. If your MultiVOIP requires service, the tech support specialist will guide you on how to send in your MultiVOIP (refer to the next section).

Contacting Technical Support

Country	By Email	By Telephone
France:	support@multitech.fr	+(33) 1-64 61 09 81
India:	support@multitechindia.com	+91 (124) 6340778
U.K.:	support@multitech.co.uk	+(44) 118 959 7774
U.S. and Canada:	support@multitech.com	(800) 972-2439
Rest of the World:	support@multitech.com	+(763) 717-5863

Recording MultiVOIP Information

Please fill in the following information on your Multi-Tech MultiVOIP. This will help tech support in answering your questions. The same information is requested on the Warranty Registration Card.

Model No.: _____
Serial No.: _____
Software Version: _____

The model and serial numbers are on the bottom of your MultiVOIP.

Please note status of your MultiVOIP including LED indicators, screen messages, diagnostic test results, problems with a specific application, and so on. Use the space below to note the MultiVOIP status:

_____

_____

_____

_____

_____

_____

_____

_____

About the Internet

If you prefer to receive technical support through the Internet, you can contact Tech Support via email at the following address:

support@multitech.com

Multi-Tech maintains a Web site at:

http://www.multitech.com

MultiVOIP™ 100

Voice / Fax over IP Networks

Appendixes

MultiTech® Systems

Appendix A - TCP/IP (Transmission Control Protocol/ Internet Protocol) Description

TCP/IP is a protocol suite and related applications developed for the U.S. Department of Defense in the 1970s and 1980s specifically to permit different types of computers to communicate and exchange information with one another. TCP/IP is currently mandated as an official U.S. Department of Defense protocol and is also widely used in the UNIX community.

Before you install TCP/IP on your network, you need to establish your Internet addressing strategy. First, choose a domain name for your company. A domain name is the unique Internet name, usually the name of your business, that identifies your company. For example, Multi-Tech's domain name is multitech.com (.com indicates this is a commercial organization; .edu denotes educational organizations, .gov denotes government organizations). Next, determine how many IP addresses you'll need. This depends on how many individual network segments you have, and how many systems on each segment need to be connected to the Internet. You will need an IP address for each network interface on each computer and hardware device.

IP addresses are 32 bits long and come in two types: network and host. Network addresses come in five classes: A, B, C, D, and E. Each class of network address is allocated a certain number of host addresses. For example, a class B network can have a maximum of 65,534 hosts, while a class C network can have only 254. The class A and B addresses have been exhausted, and the class D and E addresses are reserved for special use. Consequently, companies now seeking an Internet connection are limited to class C addresses.

Early IP implementations ran on hosts commonly interconnected by Ethernet local area networks (LAN). Every transmission on the LAN contains the local network, or medium access control (MAC), address of the source and destination nodes. The MAC address is 48-bits in length and is non-hierarchical; MAC addresses are never the same as IP addresses.

When a host needs to send a datagram to another host on the same network, the sending application must know both the IP and MAC addresses of the intended receiver. Unfortunately, the IP process may not know the MAC address of the receiver. The Address Resolution Protocol (ARP), described in RFC 826 (located at ftp://ds.internic.net/rfc/rfc826.txt) provides a mechanism for a host to determine a receiver's MAC address from the IP address. In the process, the host sends an ARP packet in a frame containing the MAC broadcast address; and then the ARP request advertises the destination IP address and asks for the associated MAC address. The station on the LAN that recognizes its own IP address will send an ARP response with its own MAC address. An ARP message is carried directly in an IP datagram.

Other address resolution procedures have also been defined, including those which allow a diskless processor to determine its IP address from its MAC address (Reverse ARP, or RARP), provides a mapping between an IP address and a frame relay virtual circuit identifier (Inverse ARP, or InARP), and provides a mapping between an IP address and ATM virtual path/channel identifiers (ATMARP).

The TCP/IP protocol suite comprises two protocols that correspond roughly to the OSI Transport and Session Layers; these protocols are called the Transmission Control Protocol and the User Datagram Protocol (UDP). Individual applications are referred to by a port identifier in TCP/UDP messages. The port identifier and IP address together form a "socket". Well-known port numbers on the server side of a connection include 20 (FTP data transfer), 21 (FTP control), 23 (Telnet), 25 (SMTP), 43 (whois), 70 (Gopher), 79 (finger), and 80 (HTTP).

TCP, described in RFC 793 (ftp://ds.internic.net/rfc/rfc793.txt) provides a virtual circuit (connection-oriented) communication service across the network. TCP includes rules for formatting messages, establishing and terminating virtual circuits, sequencing, flow control, and error correction. Most of the applications in the TCP/IP suite operate over the “reliable” transport service provided by TCP.

UDP, described in RFC 768 (ftp://ds.internic.net/rfc/rfc768.txt) provides an end-to-end datagram (connectionless) service. Some applications, such as those that involve a simple query and response, are better suited to the datagram service of UDP because there is no time lost to virtual circuit establishment and termination. UDP’s primary function is to add a port number to the IP address to provide a socket for the application.

The Application Layer protocols are examples of common TCP/IP applications and utilities, which include:

- Telnet (Telecommunication Network): a virtual terminal protocol allowing a user logged on to one TCP/IP host to access other hosts on the network, described in RFC 854 (ftp://ds.internic.net/rfc/rfc854.txt).
- FTP: the File Transfer Protocol allows a user to transfer files between local and remote host computers per IETF RFC 959 (ftp://ds.internic.net/rfc/rfc959.txt).
- Archie: a utility that allows a user to search all registered anonymous FTP sites for files on a specified topic.
- Gopher: a tool that allows users to search through data repositories using a menu-driven, hierarchical interface, with links to other sites, per RFC 1436 (ftp://ds.internic.net/rfc/rfc1436.txt).
- SMTP: the Simple Mail Transfer Protocol is the standard protocol for the exchange of electronic mail over the Internet, per IETF RFC 821 (ftp://ds.internic.net/rfc/rfc821.txt).
- HTTP: the Hypertext Transfer Protocol is the basis for exchange of information over the World Wide Web (WWW). Various versions of HTTP are in use over the Internet, with HTTP version 1.0 (per RFC 1945) (ftp://ds.internic.net/rfc/rfc1945.txt) being the most current.
- HTML: WWW pages are written in the Hypertext Markup Language (HTML), an ASCII-based, platform-independent formatting language, per IETF RFC 1866 (ftp://ds.internic.net/rfc/rfc1866.txt).
- Finger: used to determine the status of other hosts and/or users, per IETF RFC 1288 (ftp://ds.internic.net/rfc/rfc1288.txt).
- POP: the Post Office Protocol defines a simple interface between a user’s mail reader software and an electronic mail server; the current version is POP3, described in IETF RFC 1460 (ftp://ds.internic.net/rfc/rfc1460.txt).
- DNS: the Domain Name System defines the structure of Internet names and their association with IP addresses, as well as the association of mail, name, and other servers with domains.
- SNMP: the Simple Network Management Protocol defines procedures and management information databases for managing TCP/IP-based network devices. SNMP, defined by RFC 1157 (ftp://ds.internic.net/rfc/rfc1157.txt) is widely deployed in local and wide area network. SNMP Version 2 (SNMPv2), per RFC 1441< ftp://ds.internic.net/rfc/rfc1441.txt) adds security mechanisms that are missing in SNMP, but is also more complex.
- Ping: a utility that allows a user at one system to determine the status of other hosts and the latency in getting a message to that host. Ping uses ICMP Echo messages.
- Whois/NICNAME: Utilities that search databases for information about Internet domain and domain contact information, per RFC 954 (ftp://ds.internic.net/rfc/rfc954.txt).
- Traceroute: a tool that displays the route that packets will take when traveling to a remote host.

Internet Protocol (IP)

IP is the Internet standard protocol that tracks Internetwork node addresses, routes outgoing messages and recognizes incoming messages, allowing a message to cross multiple networks on the way to its final destination. The IPv6 Control Protocol (IPV6CP) is responsible for configuring, enabling, and disabling the IPv6 protocol modules on both ends of the point-to-point link. IPV6CP uses the same packet exchange mechanism as the Link Control Protocol (LCP). IPV6CP packets are not exchanged until PPP has reached the Network-Layer Protocol phase. IPV6CP packets received before this phase is reached are silently discarded. See also TCP/IP.

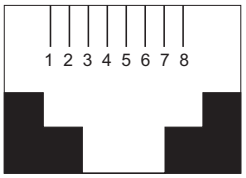
Before you install TCP/IP on your network, you need to establish your Internet addressing strategy. You first choose a domain name for your company. A domain name is the unique Internet name, usually the name of your business, that identifies your company. For example, Multi-Tech's domain name is multitech.com (where .com indicates this is a commercial organization; .edu denotes educational organizations, .gov denotes government organizations). Next, you determine how many IP addresses you'll need. This depends on how many individual network segments you have, and how many systems on each segment need to be connected to the Internet. You need an IP address for each network interface on each computer and hardware device.

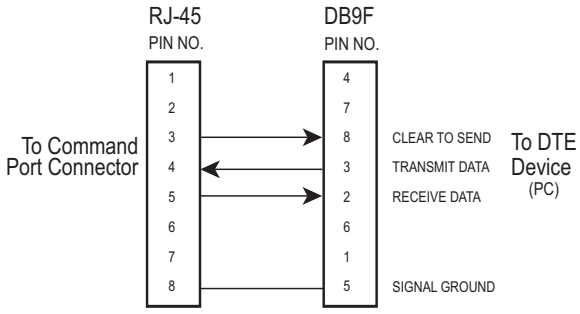
IP addresses are 32 bits long and come in two types: network and host. Network addresses come in five classes: A, B, C, D, and E. Each class of network address is allocated a certain number of host addresses. For example, a class B network can have a maximum of 65,534 hosts, while a class C network can have only 254. The class A and B addresses have been exhausted, and the class D and E addresses are reserved for special use. Consequently, companies now seeking an Internet connection are limited to class C addresses. The current demand for Internet connections will exhaust the current stock of 32-bit IP addresses. In response, Internet architects have proposed the next generation of IP addresses, Ipng (IP Next Generation). It features 16-bit addressing, surpassing the capacities of 32-bit IP.

An IP address can serve only a single physical network. Therefore, if your organization has multiple physical networks, you must make them appear as one to external users. This is done via "subnetting", a complex procedure best left to ISPs and others experienced in IP addressing. Since IP addresses and domain names have no inherent connection, they are mapped together in databases stored on Domain Name Servers (DNS). If you decide to let an Internet Service Provider (ISP) administer your DNS server, the ISP can assist you with the domain name and IP address assignment necessary to configure your company's site-specific system information. Domain names and IP addresses are granted by the InterNIC. To check the availability of a specific name or to obtain more information, call the InterNIC at (703) 742-4777.

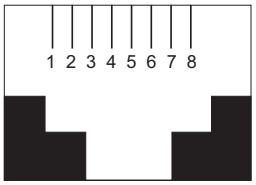
Appendix B - Cabling Diagrams

Command Port Cable



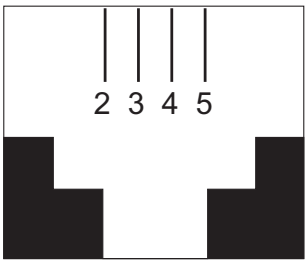


LAN Cable



Pin	Circuit Signal Name
1	TD+ Data Transmit Positive
2	TD- Data Transmit Negative
3	RD+ Data Receive Positive
6	RD- Data Receive Negative

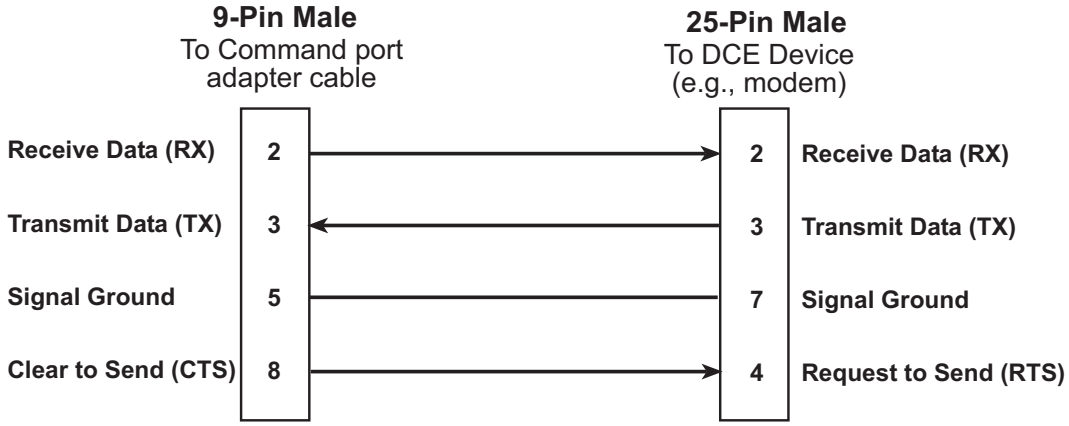
Voice/Fax Channel Connectors



Pin Connections

FXO	Description
1	N/C
2	Tip
3	Ring
4	N/C

Remote Configuration Cable



Appendix C - Regulatory Information

Class A Statement

FCC Part 15

NOTE: This equipment has been tested and found to comply with the limits for a **Class A** digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his own expense.

This device complies with Part 15 of the FCC rules.
Operation is subject to the following two conditions:

(1) This device may not cause harmful interference.

(2) This device must accept any interference that may cause undesired operation.

Warning: Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Industry Canada

This **Class A** digital apparatus meets all requirements of the Canadian Interference-Causing Equipment Regulations.

Cet appareil numerique de la classe A respecte toutes les exigences du Reglement sur le materiel brouilleur du Canada.

Fax Branding Statement

The Telephone Consumer Protection Act of 1991 makes it unlawful for any person to use a computer or other electronic device, including fax machines, to send any message unless such message clearly contains the following information:

- Date and time the message is sent
- Identification of the business or other entity, or other individual sending the message
- Phone number of the sending machine or such business, other entity, or individual

This information is to appear in a margin at the top or bottom of each transmitted page or on the first page of the transmission. (Adding this information in the margin is referred to as *fax branding*.)

Since any number of Fax software packages can be used with this product, the user must refer to the Fax software manual for setup details. Typically, the Fax branding information must be entered via the configuration menu of the software.

FCC Part 68 Telecom

1. This equipment complies with Part 68 of the Federal Communications Commission (FCC) rules. On the outside surface of this equipment is a label that contains, among other information, the FCC registration number. This information must be provided to the telephone company.
2. As indicated below, the suitable jack (Universal Service Order Code connecting arrangement) for this equipment is shown. If applicable, the facility interface codes (FIC) and service order codes (SOC) are shown.
3. An FCC-compliant telephone cord with modular plug is provided with this equipment. This equipment is designed to be connected to the phone network or premises wiring using a compatible modular jack which is Part 68 compliant. See installation instructions for details.
4. If this equipment causes harm to the phone network, the phone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the phone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.
5. The phone company may make changes in its facilities, equipment, operations, or procedures that could affect the operation of the equipment. If this happens, the phone company will provide advance notice in order for you to make necessary modifications in order to maintain uninterrupted service.
6. If trouble is experienced with this equipment (the model of which is indicated below) please contact Multi-Tech Systems, Inc., at the address shown below for details of how to have repairs made. If the equipment is causing harm to the network, the phone company may request that you remove the equipment from the network until the problem is resolved.
7. No repairs are to be made by you. Repairs are to be made only by Multi-Tech Systems or its licensees. Unauthorized repairs void registration and warranty.
8.

Manufacturer:	Multi-Tech Systems, Inc.
Trade name:	MultiVOIP
Model Numbers:	MVP120
FCC Registration Number:	AU7USA-25715-DF-N
Modular Jack (USOC):	RJ-11C or RJ-11W
Service Center in U.S.A.:	Multi-Tech Systems, Inc. 2205 Woodale Drive Mounds View, MN 55112 (763) 785-3500 Fax (763) 785-9874

Canadian Limitations Notice

Ringer Equivalence Number

Notice: The ringer equivalence number (REN) assigned to each terminal device provides an indication of the maximum number of terminals allowed to be connected to a phone interface. The termination on an interface may consist of any combination of devices subject only to the requirement that the sum of the ringer equivalence numbers of all the devices does not exceed 5.

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations. Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, phone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

CE

EMC, Safety and Terminal Directive Compliance

The CE mark is affixed to this product to confirm compliance with the following European Community Directives:

Council Directive 89/336/EEC of 3 May 1989 on the approximation of the laws of Member States relating to electromagnetic compatibility.

and

Council Directive 73/23/EEC of 19 February 1973 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits:

and

Council Directive 98/13/EC of 12 March 1998 on the approximation of the laws of Member States concerning telecommunications terminal and Satellite earth station equipment.

MultiVOIP™ 100

Voice / Fax over IP Networks

Glossary

MultiTech® Systems

Glossary

A

Access: The T1 line element made up of two pairs of wire that the phone company brings to the customer premises. The Access portion ends with a connection at the local telco (LEC or RBOC).

ACK (ACKnowledgement code) (pronounced "ack"): A communications code sent from a receiving modem to a transmitting modem to indicate that it is ready to accept data. It is also used to acknowledge the error-free receipt of transmitted data. Contrast with NAK.

Address: A numbered location inside a computer. It's how the computer accesses its resources, like a video card, serial ports, memory, etc.

Analog signal: A waveform which has amplitude, frequency and phase, and which takes on a range of values between its maximum and minimum points.

Analog Transmission: One of two types of telecommunications which uses an analog signal as a carrier of voice, data, video, etc. An analog signal becomes a carrier when it is modulated by altering its phase, amplitude and frequency to correspond with the source signal. Compare with digital transmission.

Application Program Interface (API): A software module created to allow dissimilar, or incompatible applications programs to transfer information over a communications link. APIs may be simple or complex; they are commonly required to link PC applications with mainframe programs.

ASCII (American Standard Code for Information Interchange) (pronounced "askey"): A binary code for data that is used in communications and in many computers and terminals. The code is used to represent numbers, letters, punctuation and control characters. The basic ASCII code is a 7-bit character set which defines 128 possible characters. The extended ASCII file provides 255 characters.

AT Commands: A standard set of commands used to configure various modem parameters, establish connections and disconnect. The "AT" is used to get the "attention" of the modem before the actual command is issued.

Availability: The measure of the time during which a circuit is ready for use; the complement of circuit "outage" (100% minus % outage = % available).

B

Backbone: 1. A set of nodes and their interconnecting links providing the primary data path across a network. 2. In a local area network multiple-bridge ring configuration, a high-speed link to which the rings are connected by means of bridges. A backbone can be configured as a bus or as a ring. 3. In a wide area network, a high-speed link to which nodes or data switching exchanges (DSEs) are connected. 4. A common distribution core that provides all electrical power, gases, chemicals, and other services to the sectors of an automated wager processing system.

Background: An activity that takes place in the PC while you are running another application. In other words, the active user interface does not correspond to the 'background' task.

Bandwidth: The transmission capacity of a computer channel, communications line or bus. It is expressed in cycles per second (hertz), the bandwidth being the difference between the lowest and highest frequencies transmitted. The range of usable frequencies that a transmission medium will pass without unacceptable attenuation or distortion. Bandwidth is a factor in determining the amount of information and the speed at which a medium can transmit data or other information.

Backward Explicit Congestion Notification (BECN): A bit that tells you that a certain frame on a particular logical connection has encountered heavy traffic. The bit provides notification that congestion-avoidance procedures should be initiated in the opposite direction of the received frame. See also FECN (Forward Explicit Congestion Notification).

Bell Operating Companies (BOC): The family of corporations created during the divestiture of AT&T. BOCs are independent companies which service a specific region of the US. Also called Regional Bell Operating Companies (RBOCs).

Binary Synchronous Communication (BSC): A form of telecommunication line control that uses a standard set of transmission control characters and control character sequences, for binary synchronous transmission of binary-coded data between stations.

Bit (Binary digIT): A bit is the basis of the binary number system. It can take the value of 1 or 0. Bits are generally recognized as the electrical charge generated or stored by a computer that represent some portion of usable information.

Bps (bits per second): A unit to measure the speed at which data bits can be transmitted or received. Bps differs from baud when more than one bit is represented by a single cycle of the carrier.

Buffer: A temporary storage register or Random Access Memory (RAM) used in all aspects of data communications which prevents data from being lost due to differences in transmission speed. Keyboards, serial ports, muxes and printers are a few examples of the devices that contain buffers.

Bus: A common channel between hardware devices either internally between components in a computer, or externally between stations in a communications network.

Byte: The unit of information a computer can handle at one time. The most common understanding is that a byte consists of 8 binary digits (bits), because that's what computers can handle. A byte holds the equivalent of a single character (such as the letter A).

C

Call Setup Time: The time to establish a circuit-switched call between two points. Includes dialing, wait time, and CO/long distance service movement time.

Carrier signal: An analog signal with known frequency, amplitude and phase characteristics used as a transport facility for useful information. By knowing the original characteristics, a receiver can interpret any changes as modulations, and thereby recover the information.

CCITT (Consultative Committee for International Telephone and Telegraph): An advisory committee created and controlled by the United Nations and headquartered in Geneva whose purpose is to develop and to publish recommendations for worldwide standardization of telecommunications devices. CCITT has developed modem standards that are adapted primarily by PTT (post, telephone and telegraph) organizations that operate telephone networks of countries outside of the U.S. See also ITU.

Central Office (CO): The lowest, or most basic level of switching in the PSTN (public switched telephone network). A business PABX or any residential phone connects to the PSTN at a central office.

Centrex: A multi-line service offered by operating telcos which provides, from the telco CO, functions and features comparable to those of a PBX for large business users. See also "Private Branch Exchange", "Exchange".

Channel: A data communications path between two computer devices. Can refer to a physical medium (e.g., UTP or coax), or to a specific carrier frequency.

Circuit-switched Network: A technology used by the PSTN that allocates a pair of conductors for the exclusive use of one communication path. Circuit switching allows multiple conversations on one talk path only if the end-users multiplex the signals prior to transmission.

Circuit switching: The temporary connection of two or more communications channels using a fixed, non-shareable path through the network. Users have full use of the circuit until the connection is terminated.

Client-Server: In TCP/IP, the model of interaction in distributed data processing in which a program at one site sends a request to a program at another site and awaits a response. The requesting program is called a client; the answering program is called a server.

Committed Information Rate (CIR): An agreement a customer makes to use a certain minimum data transmission rate (in bps). The CIR is part of the frame relay service monthly billing, along with actual usage, that users pay to their frame relay service provider.

Compression: 1. The process of eliminating gaps, empty fields, redundancies, and unnecessary data to shorten the length of records or blocks. 2. In SNA, the replacement of a string of up to 64-repeated characters by an encoded control byte to reduce the length of the data stream to the LU-LU session partner. The encoded control byte is followed by the character that was repeated (unless that character is the prime compression character). 3. In Data Facility Hierarchical Storage Manager, the process of moving data instead of allocated space during migration and recall in order to release unused space. 4. Contrast with decompression.

COMx Port: A serial communications port on a PC.

Congestion: A network condition where there is too much data traffic. The ITU I.233 standard defines congestion management in terms of speed and burstiness.

Congestion notification: The function in frame relay that ensures that user data transmitted at a rate higher than the CIR are allowed to slow down to the rate of the available network bandwidth.

Customer Premise Equipment (CPE): The generic term for data comm and/or terminal equipment that resides at the user site and is owned by the user with the following exclusions: Over voltage protection equipment, inside wiring, coin operated or pay telephones, "company-official" equipment, mobile phone equipment, "911" equipment, equipment necessary for the provision of communications for national defense, or multiplexing equipment used to deliver multiple channels to the customer.

D

Data Communications Equipment (DCE): Any device which serves as the portal of entry from the user equipment to a telecommunications facility. A modem is a DCE for the phone network (PSTN) that is commonly on site at the user's premises. Packet Switched Networks have another level of DCE which is most often located at a central office.

Data Service Unit (DSU): A device that provides a digital data service interface directly to the data terminal equipment. The DSU provides loop equalization, remote and local testing capabilities, and a standard EIA/CCITT interface.

Default: This is a preset value or option in software packages, or in hardware configuration, that is used unless you specify otherwise.

Device driver: Software that controls how a computer communicates with a device, such as a printer or mouse.

Digital Data: Information represented by discrete values or conditions (contrast "Analog Data").

Digital Loopback: A technique used for testing the circuitry of a communications device. Can be initiated locally, or remotely (via a telecommunications device). The tested device decodes and encodes a received test message, then echoes the message back. The results are compared with the original message to determine if corruption occurred en route.

Digital PBX: A Private Branch Exchange that operates internally on digital signals. See also "Exchange".

Digital Signal: A discrete or discontinuous signal (e.g., a sequence of voltage pulses). Digital devices, such as terminals and computers, transmit data as a series of electrical pulses which have discrete jumps rather than gradual changes.

Digital Transmission: A method of electronic information transmission common between computers and other digital devices. Analog signals are waveforms: a combination of many possible voltages. A computer's digital signal can be only "high" or "low" at any given time. Therefore, digital signals can be "cleaned up" (noise and distortion removed) and amplified during transmission.

Driver: A software module that interfaces between the Operating System and a specific hardware device (i.e. color monitors, printers, hard disks, etc.). Also known as a device driver.

Drop and Insert: The process where a portion of information carried in a transmission system is demodulated ("Dropped") at an intermediate point and different information is included ("Inserted") for subsequent transmission.

DTE (Data Terminating Equipment): A term used to include any device in a network which generates, stores or displays user information. DTE is a telecommunications term which usually refers to PCs, terminals, printers, etc.

DTMF (Dual-Tone MultiFrequency): A generic push-button concept made popular by AT&T TouchTone.

E

E&M: A telephony trunking system used for either switch-to-switch, or switch-to-network, or computer/telephone system-to-switch connection.

EIA: The Electronics Industries Association is a trade organization in Washington, DC that sets standard for use of its member companies. (See RS-232, RS-422, RS530.)

Encapsulation: A technique used by network-layer protocols in which a layer adds header information to the protocol data unit from the preceding layer. Also used in "enveloping" one protocol inside another for transmission. For example, IP inside IPX.

Endpoint: A terminal, Gateway, or MCU (MultiPoint Control Unit).

Ethernet: A 10-megabit baseband local area network that allows multiple stations to access the transmission medium at will without prior coordination, avoids contention by using carrier sense and deference, and resolves contention by using collision detection and transmission. Ethernet uses carrier sense multiple access with collision detection (CSMA/CD).

Excess Zeros: A T1 error condition that is logged when more than 15 consecutive 0s or less than one 1 bit in 16 bits occurs.

Exchange: A unit (public or private) that can consist of one or more central offices established to serve a specified area. An exchange typically has a single rate of charges (tariffs) that has previously been approved by a regulatory group.

Exchange Area: A geographical area with a single uniform set of charges (tariffs), approved by a regulatory group, for phone services. Calls between any two points within an exchange area are local calls. See also "Digital PBX", "PBX".

Exchange Termination (ET): The carrier's local exchange switch. Contrast with "Loop Termination - LT".

Explicit Congestion Management: The method used in frame relay to notify the terminal equipment that the network is overly busy. The use of FECN and BECN is called explicit congestion management. Some end-to-end protocols use FECN or BECN, but usually not both options together. With this method, a congestion condition is identified and fixed before it becomes critical. Contrast with "implicit congestion".

F

Failed Seconds: A test parameter where the circuit is unavailable for one full second.

Fax (facsimile): Refers to the bit-mapped rendition of a graphics-oriented document (fax) or to the electronic transmission of the image over phone lines (faxing). Fax transmission differs from data transmission in that the former is a bit-mapped approximation of a graphical document and, therefore, cannot be accurately interpreted according to any character code.

Firmware: A category of memory chips that hold their content without electrical power, they include ROM, PROM, EPROM and EEPROM technologies. Firmware becomes "hard software" when holding program code.

Foreground: The application program currently running on and in control of the PC screen and keyboard. The area of the screen that occupies the active window. Compare with "background".

Frequency: A characteristic of an electrical or electronic signal which describes the periodic recurrence of cycles. Frequency is inversely proportional to the wavelength or pulse width of the signal (i.e., long wavelength signals have low frequencies and short wavelength signals yield high frequencies).

Foreign Exchange (FX): A CO trunk with access to a distant CO, allowing ease of access and flat-rate calls anywhere in the foreign exchange area.

Foreign Exchange Office (FXO): provides local phone service from a CO outside of ("foreign" to) the subscriber's exchange area. In simple form, a user can pick up the phone in one city and receive a tone in the foreign city.

Connecting a POTS phone to a computer telephony system via a T1 link requires a channel bank configured for the FX connection. To generate a call from the POTS set to the computer telephony system, a FXO connection must be configured.

Foreign Exchange Station (FXS): See FX, FXO. To generate a call from the computer telephony system to the POTS set, a FXS connection must be configured.

Forward Explicit Congestion Notification (FECN): A bit that tells you that a certain frame on a particular logical connection has encountered heavy traffic. The bit provides notification that congestion-avoidance procedures should be initiated in the same direction of the received frame. See also BECN (Backward Explicit Congestion Notification).

G

Gatekeeper: An H.323 entity that provides address translation, control access, and sometimes bandwidth management to the LAN for H.323 terminals, Gateways, and MCUs.

Gateway: 1. A functional unit that interconnects two computer networks with different network architectures. A gateway connects networks or systems of different architectures. A bridge interconnects networks or systems with the same or similar architectures. 2. A network that connects hosts. 3. An H.323 entity that provides real-time, two-way communications between H.323 terminals on the LAN and other ITU terminals on a WAN, or to another H.323 Gateway.

Graphical User Interface (GUI): A type of computer interface consisting of a visual metaphor of a real-world scene, often of a desktop. Within that scene are icons, representing actual objects, that the user can access and manipulate with a pointing device.

H

H.323: An umbrella recommendation from the International Telecommunications Union (ITU) that sets standards for multimedia communications over Local Area Networks (LANs) that do not provide a guaranteed Quality of Service (QoS).

H.323 Entity: Any H.323 component, including terminals, Gateways, Gatekeepers, Multipoint Controllers (MCs), Multipoint Processors (MPs), and Multipoint Control Units (MCUs).

Handshaking: A process that two modems go through at the time of call setup to establish synchronization over the data communications link. It is a synchronization and negotiation process accomplished by the exchange of predefined, mutually recognized control codes.

High-level Data Link Control (HDLC): An ISO standard, bit-oriented data communications protocol that provides nearly error-free data transfers.

I

Hexadecimal: A base 16 numbering system used to represent binary values. Hex uses the numbers 0-9 and the letters A-F: usually notated by an "h" (e.g., "4CF h", read "four charley fox, hex"). The result is that one hex digit represents a 4-bit value.

Implicit congestion management: A method of informing the terminal that the network is busy. This method relies on the end-system protocol to detect and fix the congestion problem. (TCP/IP is an example of a protocol using only implicit congestion management.) See also "explicit congestion management".

In-band: Refers to the type of signalling over the conversion path on an ISDN call. Contrast "out-of-band".

Internet: Refers to the computer network of many millions of university, government and private users around the world. Each user has a unique Internet Address.

Internet Address (IP Address): A unique 32-bit address for a specific TCP/IP host on a network. Normally printed in dotted decimal format (e.g., 129.128.44.227).

Internet Protocol (IP): A protocol used to route data from its source to its destination in an Internet environment. The Internet Protocol was designed to connect to local area networks. Although there are many protocols that do this, IP refers to the global system of interconnecting computers. It is a highly distributed protocol (each machine only worries about sending data to the next step in the route).

Internetwork Packet Exchange (IPX): A NetWare communications protocol used to route messages from one node to another. IPX packets include network addresses and can be routed from one network to another. An IPX packet can occasionally get lost when crossing networks, thus IPX does not guarantee delivery of a complete message. Either the application has to provide that control, or NetWare's SPX protocol must be used.

Interoperable: Devices from different vendors that can exchange information using a standard's base protocol.

I/O Addresses: Locations within the I/O address space of your computer used by a device, such as an expansion card, a serial port, or an internal modem. The address is used for communication between software and a device.

IRQ Level (Interrupt Request Level): The notification a processor receives when another portion of the computer's hardware requires its attention. IRQs are numbered so that the device issuing the IRQ can be identified, and so IRQs can be prioritized.

ISA (Industry Standards Architecture) (pronounced "ice a"): The classic 8 or 16-bit architecture introduced with IBM's PC-AT computer.

ITU-TSS (formerly CCITT): International Telecommunications Union-Telecommunications Sector; the United Nations organization that prepares standards ("Recommendations") for resolving communications issues and problems.

K

Key Telephone System (KTS): Phone devices with multiple buttons that let you select incoming or outgoing CO phone lines directly. Similar in operation to a PBX, except a KTS you don't have to dial a "9" for a call outside the building.

Key Service Unit (KSU): A small device containing the switching electronics for a business key telephone system (KTS).

Key Set: A phone set with several buttons for call holding, line pickup, intercom, autodialing, etc. Also called a touchtone phone (Ericsson) and a KTS (Key Telephone Set).

L

LAPB: Link Access Procedure Balanced; based on the X.25 Layer 2 specification. A full-duplex point-to-point bit-synchronous protocol commonly used as a data link control protocol to interface X.25 DTEs. LAPB is the link initialization procedure that establishes and maintains communications between the DTE and the DCE.

LAPD: Link Access Protocol for the D-Channel; based on the ISDN Q.921 specification. A full-duplex point-to-point bit-synchronous link-level protocol for ISDN connections; different from LAPB in its framing sequence. Transmission is in units called "frames", and a frame can contain one or more X.25 packets.

Listed Directory Number (LDN): The main number assigned by the telco; the number listed in the phone directory and also provided by Directory Assistance. Some devices can have more than one LDN, such as ISDN devices that have one LDN for voice and another LDN for data.

Local Area Network (LAN): 1. A computer network located on a user's premises within a limited geographical area. Communication within a local area network is not subject to external regulations; however, communication across the LAN boundary can be subject to some form of regulation. 2. A LAN does not use store and forward techniques. 3. A network in which a set of devices are connected to one another for a communication and that can be connected to a larger network.

Local Access and Transport Area (LATA): A post-divestiture geographical area generally equivalent to a Standard Metropolitan Statistical Area. At divestiture, the territory served by the Bell system was divided into approximately 161 LATAs. The Bell Operating Companies (BOCs) provide Intra-LATA services.

Local Exchange Carrier (LEC): The local phone company which provides local (i.e., not long distance) transmission services. AKA "telco". LECs provide T1 or FT1 access to LDCs (unless the T1 circuit is completely intra-LATA). Inter-LATA T1 circuits are made up of a combination of Access and Long Haul facilities.

Local Loop: A transmission path, typically twisted-pair wire, between an individual subscriber and the nearest public telecommunications network switching center. The wires provide ISDN service, but require an NT1 at the user end and an LT at the network end. (AKA, "loop" or "subscriber loop".)

Logical Link Control (LLC2): In a local area network, the protocol that governs the exchange of transmission frames between data stations independently of how the transmission medium is shared. The LLC2 protocol was developed by the IEEE 802 committee and is common to all LAN standards.

Logical Unit (LU): A type of network accessible unit that enables end users to gain access to network resources and communicate with each other.

M

Management Information Base (MIB): A database of network management information used by the Common Management Information Protocol (CMIP) and the Simple Network Management Protocol (SNMP).

Modem: A communications device that enables a computer to transmit information over a phone line. It converts the computer's digital signals into analog signals to send over a phone line and converts them back to digital signals at the receiving end. Modems can be internal and fit into an expansion slot, or external and connect to a serial port.

Multiplexer (Mux): 1. A device that takes several input signals and combines them into a single output signal in such a manner that each of the input signals can be recovered. 2. A device capable of interleaving the events of two or more activities or capable of distributing the events of an interleaved sequence to the respective activities. 3. Putting multiple signals on a single channel.

Multiprotocol: A device that can interoperate with devices utilizing different network protocols.

Multithreading: The ability of a software system to be able to handle more than one transaction concurrently. This is contrasted to the case where a single transaction is accepted and completely processed before the next transaction processing is started.

N

NAK (Negative Acknowledgment): Communications code used to indicate that a message was not properly received, or that a terminal does not wish to transmit. Contrast with ACK.

Network: A group of computers connected by cables or other means and using software that enables them to share equipment, such as printers and disk drives to exchange information.

Node: Any point within a network which has been assigned an address.

O

Object-Orientated: A method for structuring programs as hierarchically organized classes describing the data and operations of objects that can interact with other objects.

Off-hook: The condition of a device which has accessed a phone line (with or without using the line). In modem use, this is equivalent to a phone handset being picked up. Dialing and transmission are allowed, but incoming calls are not answered. Contrast "on-hook".

Off Premise Extension (OPX): An extension or phone that terminates in a location other than that of the PBX. Commonly used to provide a corporate member with an extension of the PBX at home.

On-Hook: The condition of a device which has not accessed a phone line. In modem use, this is equivalent to a phone handset that has not been picked up. In other words, it can receive an incoming call. Contrast "off-hook".

Open Shortest Path First (OSPF): A hierarchical Interior Gateway Protocol (IGP) routing algorithm for IP that is a proposed standard for Internet. OSPF incorporates least-cost routing, equal-cost routing, and load balancing.

Outage: The measure of the time during which a circuit is not available for use due to service interrupt. Outage is the complement of circuit "availability" (100% minus % available = % outage).

Out-of-band: Signaling that is separated from the channel carrying the information (i.e., the voice/data/video signal is separate from the carrier signal). Dialing and various other "supervisory" signals are included in the signaling element. Contrast "In-band" signaling.

Out of Frame (OOF): A T1 alarm condition that is logged on the loss of 2, 3 or 4 of 5 consecutive FT framing bits.

P

Packet: 1. In data communication, a sequence of binary digits, including data and control signals, that is transmitted and switched as a composite whole. The data, control signals and, possibly, error control information are arranged in a specific format. 2. Synonymous with data frame. 3. In TCP/IP, the unit of data passed across the interface between the Internet layer and the link layer. A packet includes an IP header and data. A packet can be a complete IP datagram or a fragment of an IP diagram. 4. In X.25, a data transmission information unit. A group of data and control characters, transferred as a unit, determined by the process of transmission. Commonly used data field lengths in packets are 128 or 256 bytes. 5. The field structure and format defined in the CCITT X.25 recommendation.

Packet Data: The information format ("packetized") used for packet-mode calls.

Packet Mode: Refers to the switching of chunks of information for different users using statistical multiplexing to send them over the same transmission facility.

Parity bit: An extra bit attached to each byte of synchronous data used to detect errors in transmission.

Physical Unit (PU): The component that manages and monitors the resources (such as attached links and adjacent link stations) associated with a node, as requested by an SSCP via an SSCP-PU session. An SSCP activates a session with the physical unit in order to indirectly manage, through the PU, resources of the node such as attached links. This term applies to type 2.0, type 4, and type 5 nodes only.

Point of Presence (POP): The central office's end points of the long distance carriers.

Point to Point Protocol (PPP): A protocol that lets a PC user access TCP/IP (Internet member) using an ISDN terminal adapter or a high-speed modem over a standard phone line.

Port: A location for input or output data exchange. Computers, muxes, etc. have ports for various purposes.

Primitive: An abstract representation of interaction across the access points indicating that information is being passed between the service user and the service provider. The OSI Reference Model defines four types of primitives: Request, Indication, Response and Confirm.

Private Branch Exchange (PBX): A phone exchange located on the customer's premises. The PBX provides a circuit switching facility for phone extension lines within the building, and access to the public phone network. See also "Exchange".

PROM (Programmable Read Only Memory - pronounced "prom"): A permanent memory chip that can be programmed or filled by the customer after by the manufacturer has set initial values. Contrast with ROM.

Protocol: 1. A set of semantic and syntactic rules that determines the behavior of functional units in achieving communication. 2. In Open Systems Interconnection architecture, a set of semantic and syntactic rules that determine the behavior of entities in the same layer in performing communication functions. 3. In SNA, the meanings of and the sequencing rules for requests and responses used for managing the network, transferring data, and synchronizing the states of network components. 4. Synonymous with line control discipline.

PSTN (Public Switched Telephone Network): A worldwide public voice phone network that is used as a telecommunications medium for the transmission of voice, data and other information.

Public Data Network (PDN): A packet-switched network that is available to the public for individual ("subscriber") use. Typically, controlled by a government or a national monopoly.

Public Switched Telephone Network (PSTN): The group of circuit-switching voice carriers, which are commonly used as analog data communications services.

Pulse dialing: One of two methods of dialing a phone, usually associated with rotary-dial phones. Compare with "tone dialing".

Q

Quality of Service (QoS): Guarantees network bandwidth and availability for applications.

Quantizing: The process of analog-to-digital conversion by assigning a range, from the contiguous analog values, to a discrete number.

R

Random Access Memory (RAM): A computer's primary workspace. All data must be stored in RAM (even for a short while), before software can use the processor to manipulate the data. Before a PC can do anything useful it must move programs from disk to RAM. When you turn it off, all information in RAM is lost.

RAS Channel: An unreliable channel used to convey the Registration, Admissions and Status messages and bandwidth changes between two H.323 entities.

Recognized Private Operating Agency (RPOA): A corporation, private or government-controlled, that provides telecommunications services. RPOAs, such as AT&T, participate as non-voting members in the CCITT.

Request for Comment (RFC): A set of papers in which Internet standards (published and proposed), along with generally-accepted ideas, proposals, research results, etc. are published.

RJ-11: An industry standard interface used for connecting a phone to a modular wall outlet; comes in 4-and 6-wire packages.

RJ-45: An 8-wire modular connector for voice and data circuits.

Router: A device that connects two networks using the same networking protocol. It operates at the Network Layer (Layer 3) of the OSI model for forwarding decisions.

Routing Information Protocol (RIP): A distance vector-based protocol that provides a measure of distance, or hops, from a transmitting workstation to a receiving workstation.

RS-232-C: An EIA standard for a serial interface between computers and peripheral devices (modem, mouse, etc.). It uses a 25-pin DB-25, or a 9-pin DB-9 connector. The RS-232 standard defines the purposes, electrical characteristics and timing of the signals for each of the 25 lines.

RS-422: The EIA standard for a balanced interface with no accompanying physical connector. RS-422 products can use screw terminals, DB-9, various DB-25, and DB-37 connectors.

RS-530: The EIA standard for the mechanical/electrical interface between DCEs and DTEs transmitting synchronous or asynchronous serial binary data. RS-530 provides for high data rates with the same connector used for RS-232; however, it is incompatible with RS-232.

S

Serial Port: The connector on a PC used to attach serial devices (those that need to receive data one bit after another), such as a mouse, a printer or a modem. This consists of a 9- or 25-pin connector that sends data in sequence (bit by bit). Serial ports are referred to as "COMx" ports, where x is 1 to 4 (i.e., COM1 through COM4). A serial port contains a conversion chip called a "UART" which translates between internal parallel and external serial formats.

Service: The requirements offered by an RPOA to its customers to satisfy specific telecommunications needs.

Signaling: The process of establishing, maintaining, accounting for, and terminating a connection between two endpoints (e.g., the user premises and the telco CO). Central office signals to the user premises can include ringing, dial tone, speech signals, etc. Signals from the user's phone can include off-hook, dialing, speech to far-end party, and on-hook signals.

In-band signaling techniques include pulse and tone dialing. With common channel signaling, information is carried out-of-band.

Simple Network Management Protocol (SNMP): TCP/IP protocol that allows network management.

Simultaneous Voice Data (SVD): A technology for letting a user send data via a modem, and use a handset to talk to another user at the same time over the same connection. The alternative, making a second call, can be expensive or even impossible. The uses for SVD are telecommuting, videoconferencing, distant learning, tech support, etc.

Stop Bit: One of the variables used for timing in asynchronous data transmission. Depending on the devices, each character may be trailed by 1, 1.5, or 2 stop bits.

Subscriber Loop: See "Local loop".

Switched 56: A circuit-switched (full duplex digital synchronous data transmission) service that lets you dial a number and transmit data to it at 56K bps. It is a relatively low cost service, widely used in North America for telecommuting, videoconferencing and high speed data transfers. Many phone companies are (or will be) phasing out Switched 56 in favor of ISDN service.

Switched Virtual Circuit (SVC): A type of data transmission where the connection is maintained only until the call is cleared.

Switched Line: In communications, a physical channel established by dynamically connecting one or more discreet segments. This connection lasts for the duration of the call after which each segment can be used as part of a different channel. Contrast with leased line.

Switched Network: A network in which a temporary connection is established from one point via one or more segments.

Synchronous Transmission: The transmission of data which involves sending a group of characters in a packet. This is a common method of transmission between computers on a network or between modems. One or more synchronous characters are transmitted to confirm clocking before each packet of data is transmitted. Compare to Asynchronous Transmission.

Systems Network Architecture (SNA): The description of the logical structure, formats, protocols, and operational sequences for transmitting information units through, and controlling the configuration and operation of , networks.

T

Tariff: The rate/availability schedule for telephone and ISDN services from a regulated service provider.

TCP/IP: A set of communication protocols that support peer-to-peer connectivity functions for both local and wide area networks.

Transmission Control Protocol / Internet Program (TCP/IP): A multilayer set of protocols developed by the US Department of Defense to link dissimilar computers across dissimilar and unreliable LANs.

Terminal: The screen and keyboard device used in a mainframe environment for interactive data entry. Terminals have no "box", which is to say they have no file storage or processing capabilities.

Tie line: A dedicated circuit linking two points without having to dial a phone number (i.e., the line can be accessed by lifting the phone handset or by pushing a button).

Toll Call: A call to a location outside of your local service area (i.e., a long distance call).

Tone dialing: One of two methods of dialing a phone, usually associated with Touch-Tone® (push button) phones. Compare with pulse dialing.

Topology: Physical layout of network components (cables, stations, gateways, and hubs). Three basic interconnection topologies are star, ring, and bus networks.

Transmission Control Protocol (TCP): A communications protocol used in Internet and in any network that follows the US Department of Defense standards for internetwork protocol. TCP provides a reliable host-to-host protocol between hosts in packet-switched communications networks and in interconnected systems of such networks. It assumes that the Internet protocol is the underlying protocol.

Trunk: Transmission links that interconnect switching offices.

Tunneling: Encapsulation data in an IP packet for transport across the Internet.

Twisted pair wiring: A type of cabling with one or more pairs of insulated wires wrapped around each other. An inexpensive wiring method used for LAN and telephone applications, also called UTP wiring.

U

UART (Universal Asynchronous Receiver/Transmitter) (pronounced "you art"): A chip that transmits and receives data on the serial port. It converts bytes into serial bits for transmission, and vice versa, and generates and strips the start and stop bits appended to each character.

UNIX: An operating system developed by Bell Laboratories that features multiprogramming in a multi-user environment.

Unshielded Twisted Pair (UTP): Telephone-type wiring. Transmission media for 10Base-T.

V

V.25bis: An ITU-T standard for synchronous communications between a mainframe or host and a modem using HDLC or other character-oriented protocol.

V.54: The ITU-T standard for local and remote loopback tests in modems, DCEs and DTEs. The four basic tests are:
- local digital loopback (tests DTE send and receive circuits),
- local analog loopback (tests local modem operation),
- remote analog loopback (tests comm link to the remote modem), and
- remote digital loopback (tests remote modem operation).

Virtual Circuit: A logical connection. Used in packet switching wherein a logical connection is established between two devices at the start of transmission. All information packets follow the same route and arrive in sequence (but do not necessarily carry a complete address).

Z

Zone: A collection of all terminals, Gateways, and MCUs managed by a single Gatekeeper. A zone must include at least one terminal and can include LAN segments connected using routers.

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